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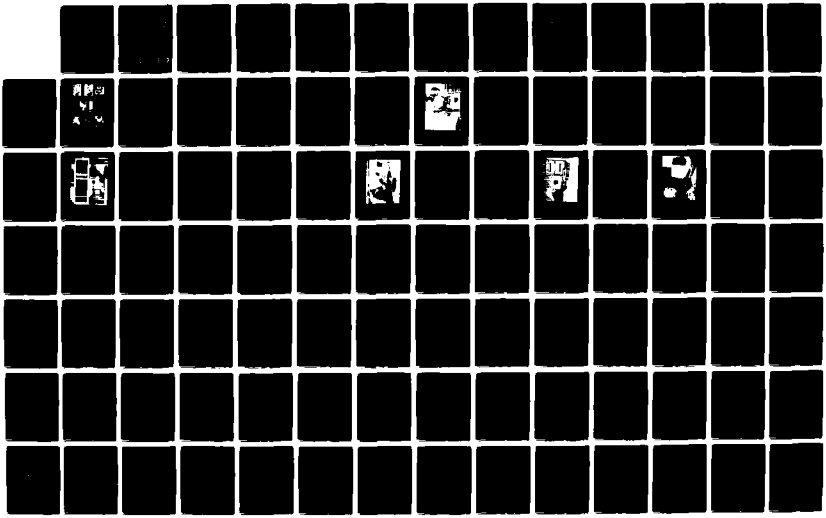

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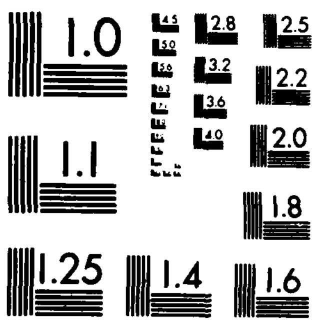

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Hardware Description of Mass Weather Dissemination System Exploratory Engineering Model

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September 1982

Final Report

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16. Abstract

This report describes the Mass Weather Dissemination System Exploratory Engineering Model currently being tested in the Flight Service Station Engineering Laboratory. The object of this effort is to investigate through development, test, and evaluation, the application of digital technology to the mass dissemination of meteorological and aeronautical information. The prototype model is a fully automated system designed to transfer a significant amount of workload from the flight service specialist to system hardware and to provide better service to the flying public.

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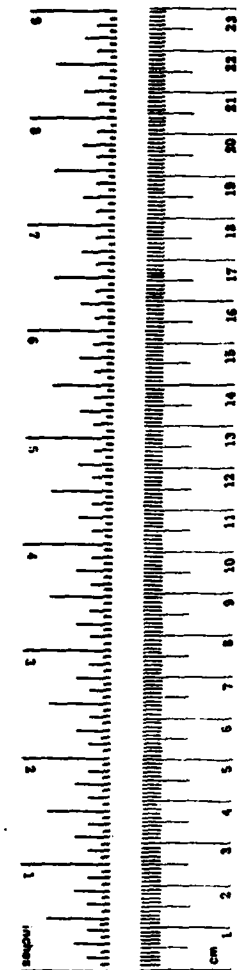
METRIC CONVERSION FACTORS

Approximate Conversions to Metric Measures

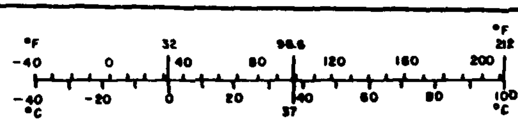
Symbol	When You Know	Multiply by	To Find	Symbol
		LENGTH		
in	inches	*2.5	centimeters	cm
ft	feet	30	centimeters	cm
yd	yards	0.9	meters	m
mi	miles	1.6	kilometers	km
		AREA		
in^2	square inches	6.5	square centimeters	cm^2
ft^2	square feet	0.09	square meters	m^2
yd^2	square yards	0.8	square meters	m^2
mi^2	square miles	2.6	square kilometers	km^2
	acres	0.4	hectares	ha
		MASS (weight)		
oz	ounces	28	grams	g
lb	pounds	0.45	kilograms	kg
	short tons (2000 lb)	0.9	tonnes	t
		VOLUME		
tsp	teaspoons	5	milliliters	ml
Tbsp	tablespoons	15	milliliters	ml
fl oz	fluid ounces	30	milliliters	ml
c	cups	0.24	liters	l
pt	pints	0.47	liters	l
qt	quarts	0.95	liters	l
gal	gallons	3.8	liters	l
ft^3	cubic feet	0.03	cubic meters	m^3
yd^3	cubic yards	0.76	cubic meters	m^3
		TEMPERATURE (exact)		
°F	Fahrenheit temperature	5/9 (after subtracting 32)	Celsius temperature	°C

*1 in ≡ 2.54 (exactly). For other exact conversions and more detailed tables, see NBS Misc. Publ. 286, Units of Weights and Measures, Price $2.25, SD Catalog No. C13.10:286.



Approximate Conversions from Metric Measures

Symbol	When You Know	Multiply by	To Find	Symbol
		LENGTH		
mm	millimeters	0.04	inches	in
cm	centimeters	0.4	inches	in
m	meters	3.3	feet	ft
m	meters	1.1	yards	yd
km	kilometers	0.6	miles	mi
		AREA		
cm^2	square centimeters	0.16	square inches	in^2
m^2	square meters	1.2	square yards	yd^2
km^2	square kilometers	0.4	square miles	mi^2
ha	hectares (10,000 m^2)	2.5	acres	
		MASS (weight)		
g	grams	0.035	ounces	oz
kg	kilograms	2.2	pounds	lb
t	tonnes (1000 kg)	1.1	short tons	
		VOLUME		
ml	milliliters	0.03	fluid ounces	fl oz
l	liters	2.1	pints	pt
l	liters	1.06	quarts	qt
l	liters	0.26	gallons	gal
m^3	cubic meters	35	cubic feet	ft^3
m^3	cubic meters	1.3	cubic yards	yd^3
		TEMPERATURE (exact)		
°C	Celsius temperature	9/5 (then add 32)	Fahrenheit temperature	°F



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INTRODUCTION

PURPOSE.

The purpose of this project was to investigate through development, test, and evaluation, the application of digital technology to the mass dissemination of meteorological and aeronautical information. This report describes the hardware of the Federal Aviation Administration (FAA) Technical Center Mass Weather Dissemination System Engineering Model. A software description is available in report DOT/FAA/RD-81/1.

The report contains a list of design objectives, a general block diagram description of the Engineering Model developed in-house, and an example of a pilot's interaction with the system.

Detailed descriptions of the subsystems of the Engineering Model, which were also developed in-house, appear in the appendices.

BACKGROUND.

The Department of Transportation (DOT) study "A Proposal for the Future of Flight Service Stations," dated August 1973, recommended that greater emphasis be placed on the mass dissemination of aviation weather briefings and that the one-to-one method of briefing be reduced or eliminated: "There is a need to reduce or eliminate the one-to-one method of briefing, particularly for heavily traveled routes or repetitive briefings. The function lends itself well to automation. Therefore, automation (self-briefing) and mass dissemination should be emphasized."

In response to this proposal, the FAA set out to determine whether or not an improved Pilots Automated Telephone Weather Answering Service (PATWAS) system would result in transferring workload from specialist to hardware as projected in the DOT study. This resulted in the New York City PATWAS test, a joint FAA/National Weather Service (NWS) experimental effort centered at the NWS Office, LaGuardia Airport, New York.

The improved New York City PATWAS equipment provided pilots with three different weather briefings. These briefings consisted of a generally northbound route-oriented briefing, a generally southbound route-oriented briefing, and a local area briefing. Each message was accessed by dialing a separate telephone number. Figure 1 is a functional description of this briefing system.

The telephone connection equipment permitted up to 10 pilots to be connected to each message. A pilot calling a particular message would be connected to the message by the barge-in method. Barge-in is a connection arrangement by which the first pilot receives the message from the beginning, and subsequent pilots are connected to the message at the point in the message that is currently being played-out to the first pilot.

Each briefing was recorded on a special tape recorder that allowed briefings to be recorded in five segments by means of cassette cartridges. When one cartridge came to its end, the next cartridge was started. Thus, a briefing was played repetitively until pilots were no longer connected to that briefing.

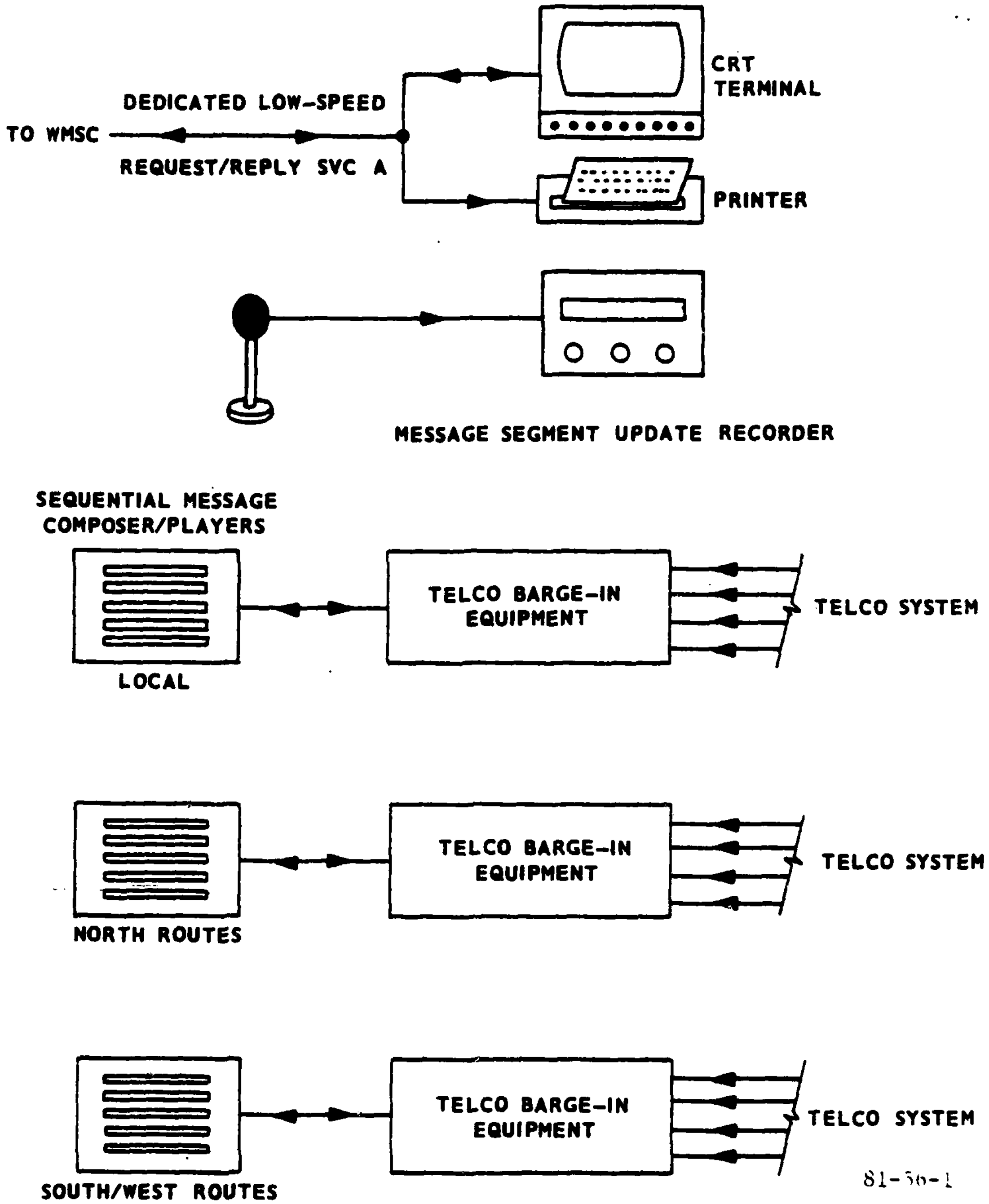

FIGURE 1. FUNCTIONAL DESCRIPTION OF THE TRIAL PATWAS OPERATION

The New York City PATWAS test ran for approximately 1 year during which time subjective and objective data were systematically collected. Data collected consisted of (1) briefing activity for 5 years preceding the test and during the test, (2) PATWAS activity data for 5 years preceding the test for the old PATWAS system and during the test for both the old and improved PATWAS systems, and (3) questionnaire data from four questionnaires, one for an initial appraisal and three for subsequent appraisals of the improved PATWAS system.

An extensive analysis of the data demonstrated conclusively that the premise offered in the DOT study was valid. Further, it was determined that a pilot occupied on the average 50 percent less of a briefer's time after having listened to a PATWAS briefing.

In September 1976, the FAA issued a report on the General Aviation activity for the year 1975. Certain interesting statistics extracted from this report are:

1. Approximately 71 percent of the flights are local. The remaining 29 percent are itinerant.

2. The average time per local flight is 57 minutes.

3. The average itinerant flight takes 76 minutes over a distance of 186 miles.

4. Approximately 67 percent of the pilots do not file flight plans; of the remaining 33 percent, 14 percent file instrument flight rules (IFR).

5. Approximately 8 percent of local and 54 percent of those flying cross country file flight plans.

Conclusions and estimates drawn from the New York City PATWAS test and the General Aviation activity studies are:

1. No recorded briefing system can replace the briefer.

2. Recorded, current, accessible briefings will transfer workload from the briefer to hardware.

3. Pilots having listened to a PATWAS briefing upon calling a briefer occupy 50 percent less of the briefer's time.

4. An estimated 75 percent of General Aviation flights are within 200 miles of the point of departure, with 52 percent being local flights.

5. A local area and selected general route-oriented briefings can serve the majority of flights most of the time.

6. One-call/one-number complete service could promote safety through increased use of the system.

Although the improved PATWAS system was much better than the old PATWAS system, even the improved system had three major deficiencies. One deficiency was fragmented service. A pilot had to call one or two numbers for his briefing and another one to file his flight plan or consult with a specialist.

A second deficiency was the barge-in connection method. Briefings were prepared in a logical way so that the bad news came first. Most pilots could decide from this introductory information that they could not fly; they would then hang up, thus, reducing line hold time. The barge-in method of connecting a pilot to the briefing enables only the first pilot to access the system to listen to the briefing from the beginning. All subsequent pilots to access the briefing must listen to the briefing from some intermediate point. These pilots could listen to the beginning of the briefing message by allowing the briefing to cycle and start playing from the beginning again, but only at a cost of increased line hold time.

A third deficiency of the improved PATWAS system was poor telephone line utilization. Frequently, all 10 telephone lines servicing the local briefing were being used. Additional pilots seeking local briefings were given a busy signal even though all of the lines on the north and south briefings were not being used.

If the conclusions and estimations from the New York City PATWAS test and the General Aviation activity studies were reasonable, the question of how to implement them in a reasonable way had to be answered. The answer that presented itself was to automate, where feasible, through the application of digital technology; thus, the Mass Weather Dissemination System Exploratory Engineering Model project was established as the vehicle to investigate the potential benefits of implementation.

DESIGN OBJECTIVES.

The design objectives of the engineering model were to provide one-number complete service, to improve service to the pilot, to reduce specialists' workload, and to provide operational economy in the dissemination of meteorological and aeronautical information to the pilot population. These improvements were based on the service provided by the improved New York City PATWAS system (described in the Background discussion) and the old PATWAS system. In brief terms, the old PATWAS system was a single tape recorded briefing that played continuously to several telephone lines simultaneously. The number of lines serviced by this barge-in connection method was dependent upon the location of the equipment.

One-number complete service means that a pilot, by dialing only one telephone number, could listen to any of five prerecorded briefings, communicate with a flight service specialist, or access the fast-file recorder and, thus, file, amend, or close a flight plan. Previously, the pilot had to dial one or several numbers to receive his briefing and a separate number to access the fast-file recorder. If the pilot required communication with a flight service specialist, he had to dial yet another number.

Service is improved to the pilot because he can receive all of his required services from just one telephone call. The most significant weather information is presented at the beginning of the message, thus giving the pilot the ability to make a go/no-go decision early in the briefing. The pilot is not required to have special telephone equipment to access the system; any telephone instrument will operate the system. The final improvement from the pilot's point of view is the ability to review information that he has just spoken into the fast-file recorder. If he detects that he has made an error, he is given the opportunity to make an immediate amendment.

Specialist workload is reduced by increasing the efficiency by which he enters messages to the recorded briefing system and by partially automating the flight plan filing process. Each element of the recorded briefing is individually accessible by the specialist. When the information for a message element changes, only that small segment is changed. The remainder of the briefing is not modified. The specialist may review at any time any briefing in its entirety or any message element.

The flight plan entry position allows the specialist to enter flight plan data in a random manner as the pilot recorded it on the fast-file recorder. The rigid formatting of the flight plan is accomplished by the computer.

One other feature of the system that will eventually reduce specialist workload i called Automatic Message Composition. Automatic Message Composition takes textual message and converts it into speech. As each word is received in te form, its speech representation is retrieved from bulk storage and placed in buffer. A complete voice message is available for dissemination after th receipt of the last word of text. At this point, this new voice message may disseminated or may be reviewed by a specialist and then disseminated. This pa of the Mass Weather Dissemination System is a separate project currently unde development and will be fully discussed in a separate report.

With Automatic Message Composition, the specialist reviews messages rather than manually entering them. He will only manually enter messages when the message communication line is out of service or to enter emergency reports.

Operational economy is achieved by reducing telephone line costs. In this system, there are no dedicated lines for specific functions. Fewer lines are required since all lines are dynamically assigned to different services as specified by the pilot. Since the most significant weather information is placed at the beginning of the message, and the message is presented to the pilot from the beginning, there is a chance that the pilot's line hold time will be significantly reduced. Thus, a given number of pilots may receive the same level of service with fewer telephone lines and, consequently, reduced communications cost.

ENGINEERING MODEL SPECIFICATIONS.

The design of the Engineering Model was controlled by a basic set of specifications. These specifications were an attempt to balance implementation with operational practicality and operational economy.

The Engineering Model will provide one-number complete service to the pilot population that it serves. By dialing only one number, the pilot will have the ability to select and listen to one or several prerecorded briefings; communicate with a flight service specialist; or file, amend, or close-out a flight plan via a fast-file recorder. Simultaneous service for up to 20 telephone lines will be provided.

The Engineering Model will provide the ability to play any combination of five prerecorded briefings to 20 telephone lines simultaneously (message to line-multiplexing). Each of the five prerecorded briefings will have a maximum playback time of 10 minutes duration. Each briefing that is selected will be played-out from its beginning (synchronous access). Message update will be transparent to pilots presently accessing the system.

System configuration shall be pilot controlled by voice commands. Such a voice recognition device must be speaker independent, must operate over switched telecommunications lines, and must service all 20 telephone lines.

System prompts and instructions for system use, which are presented to the pilot, will appear in detailed and abbreviated form. Pilots who are familiar with using the system receive the abbreviated instructions, while pilots who are not familiar with using the system may elect to receive detailed instructions.

The Engineering Model will provide four specialist answering sets and two fast-file recorders. Any of the 20 telephone lines will have access to any of these devices. All switching of telephone lines will be done internally within the system.

The system will be easily operated by people who have little or no computer experience.

ENGINEERING MODEL FUNCTIONAL DESCRIPTION

The Mass Weather Dissemination System Exploratory Engineering Model consists of 13 system components as shown in figure 2. Figure 3 is a pictorial diagram of the engineering model. The system components are the Interdata 7/32 computer, voice digitizer, message entry console, bulk storage subsystem, telephone voice output channel, utterance recognition device (URD), telephone terminal equipment, telephone line controller, specialist answering set, fast-file recorder, flight plan entry console, printer, and automatic message entry console.

INTERDATA 7/32 COMPUTER.

The Perkin-Elmer 7/32 computer performs system integration and control of the entire Mass Weather Dissemination System. Operating under stored program control, it responds to requests for service from all system components. A block diagram of the 7/32 computer is shown in figure 4.

The 7/32 directs the URD to listen to the pilot on any one of the 20 telephone lines. It then responds to encoded responses from the URD to connect, disconnect, or playback the fast-file recorder; to connect or disconnect a specialist answering set; or to select and playback any of five prerecorded briefings to the pilot on one of the telephone lines. The 7/32 computer also services the message entry console and assigns digitized voice messages to appropriate locations in the bulk storage. Also serviced is the flight plan entry position, where flight plans from fast-file are entered in random format, edited, and reformatted to Service B transmission requirements and then are transmitted to the printer. All of these transactions are going on concurrently. The Mass Weather Dissemination System requires a general purpose processor that can efficiently transfer large quantitites of data from one point to another. The 7/32 computer performs this task well.

The easy-to-use and convenient real-time multitasking operating system that was supplied with the 7/32 computer also contributed to the 7/32 computer's applicability to the Mass Weather Dissemination System's operation.

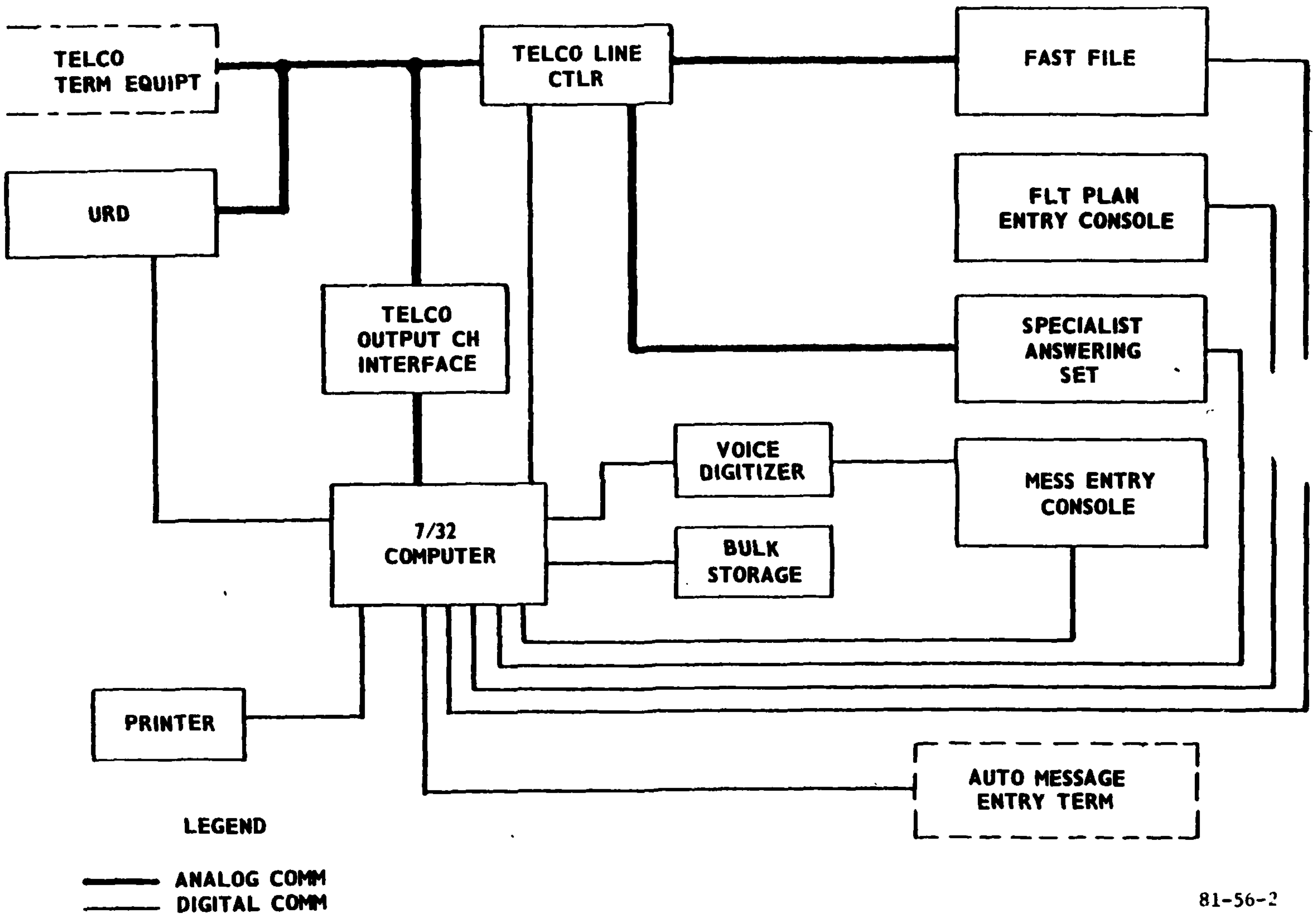

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FIGURE 2. MASS WEATHER DISSEMINATION EXPLORATORY ENGINEERING MODEL (SCHMATIC)

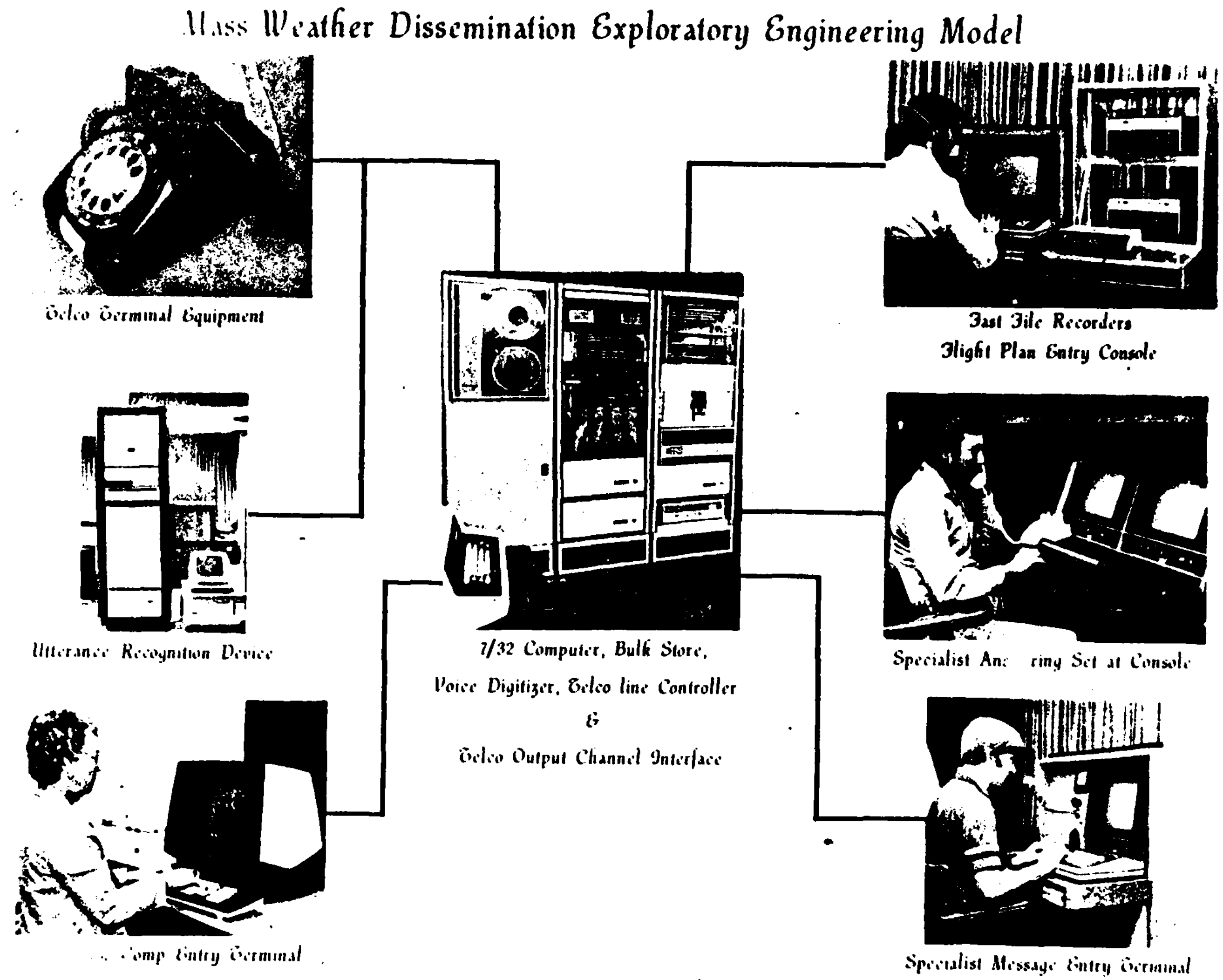


FIGURE 3. MASS WEATHER DISSEMINATION EXPLORATORY ENGINEERING MODEL

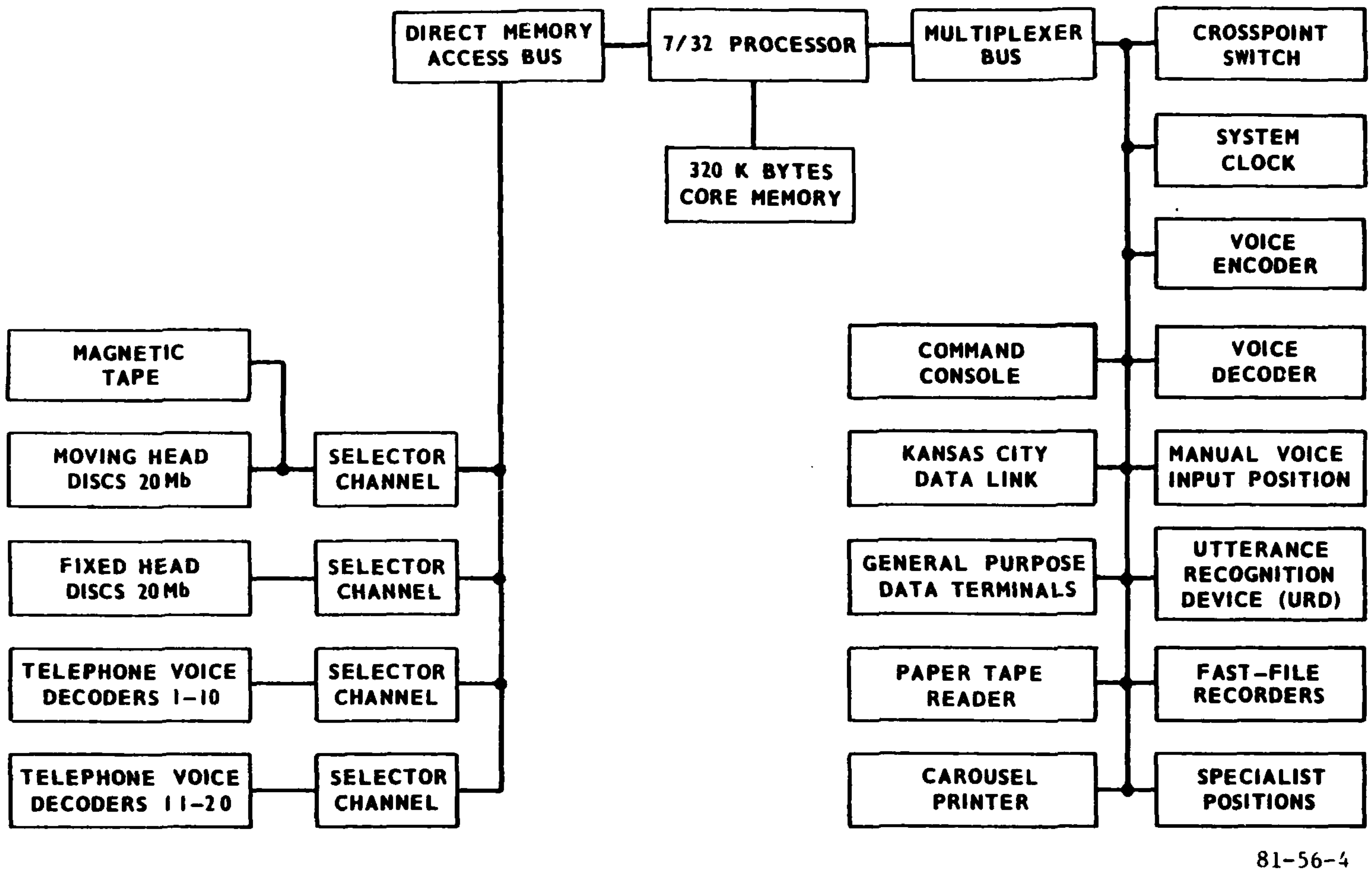


FIGURE 4. 7/32 COMPUTER SYSTEM BLOCK DIAGRAM

VOICE DIGITIZER.

The Voice Digitizer consists of the voice encoder and voice decoder. The voice encoder converts the analog voice signal from a microphone which is located at the message entry console into encoded digital form for storage in the bulk storage. The voice decoder converts digitized speech from the bulk storage into analog speech form. This analog speech is amplified and played out via a loudspeaker which is located at the message entry console.

VOICE ENCODER. The voice encoder takes an analog voice signal, filters the signal, performs analog-to-digital conversions on the filtered signal, encodes the digital voice samples, and presents the encoded digital voice data to the 7/32 processor for storage and subsequent retrieval for dissemination (figure 5). A detailed description of the voice encoder is available in appendix A.

The audio input is first passed through the input level control circuitry to insure that the audio input to the analog-to-digital converter is within operational limits.

The level controlled audio signal is next passed through the anti-aliasing filter. The anti-aliasing filter removes the high frequency components of the speech signal to prevent an unwanted phenomenon which occurs when the sample rate of the analog-to-digital converter and the analog signal frequency are similar.

The output of the anti-aliasing filter feeds the sample-and-hold amplifier. The sample-and-hold amplifier maintains the analog signal at a constant value during the short time that the analog-to-digital converter is actually making a conversion.

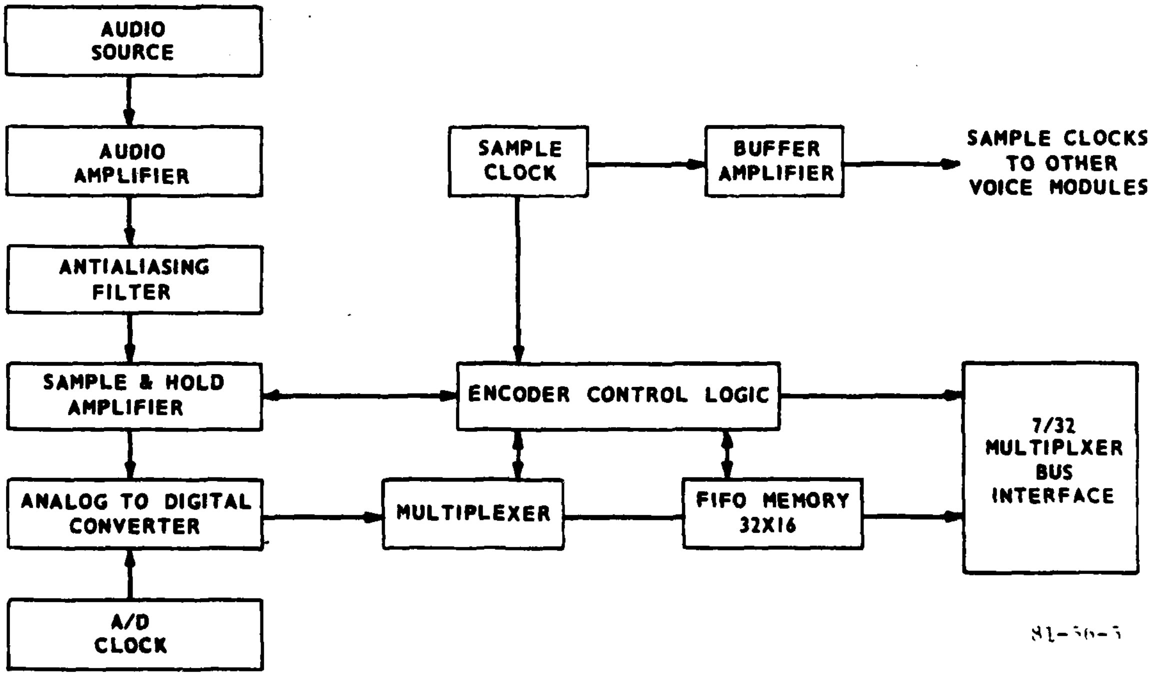


FIGURE 5. VOICE ENCODER BLOCK DIAGRAM

The heart of the voice encoder is a companding analog-to-digital converter. The companding analog-to-digital converter converts the analog voice signal into digital voice samples at a rate determined by the sample clock.

Several voice samples are collected to form a 7/32 halfword so as to efficiently utilize the processor's memory. This task is performed by the multiplexer.

The memory is a first-in-first-out (FIFO) memory located on the voice encoder module that prevents the potential loss of digitized voice data when the 7/32 processor is temporarily too busy to respond to the encoder module's data available request. At any time that data are available in the encoder module's FIFO memory, the processor is interrupted via the processor interface control logic. Interrupts continue until the encoder module's FIFO memory contains no more digitized voice data.

The voice sample clock determines the sample rate of the voice encoder and is distributed to all other voice boards in the system. The voice sample clock was set to give 6,000 voice samples per second for this system.

VOICE DECODER. The voice decoder takes digitally encoded voice data from the 7/32 processor, converts the digital voice samples into a speech signal, low-pass filters the speech signal, and then amplifies the speech signal and plays it into a loudspeaker (figure 6). A detailed description of the voice decoder is available in appendix B.

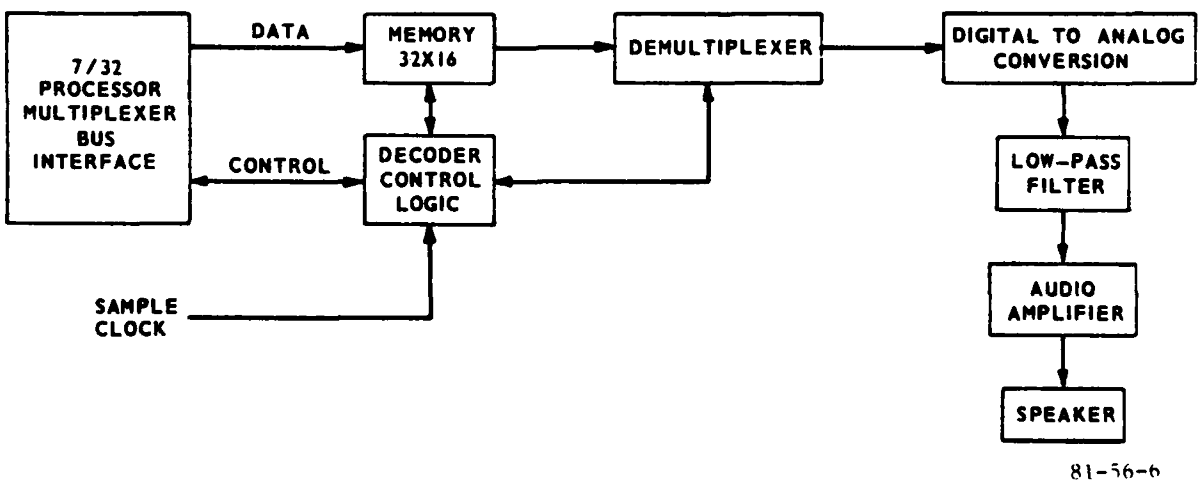


FIGURE 6. VOICE DECODER BLOCK DIAGRAM

Data from the 7/32 processor are presented to the voice decoder via the processor interface. The 16-bit data words are stored in the voice decoder's internal memory.

The voice decoder's memory is a FIFO memory that prevents the potential stoppage of speech playback when the 7/32 processor is temporarily too busy to respond to the voice decoder's data request signals. Whenever the FIFO memory has a vacant position, it interrupts the 7/32 processor to fill that vacancy. Interrupts continue until the memory is full.

Five-bit voice samples are separated from the 16-bit voice data words by the demultiplexer at a rate determined by the sample clock (6,000 hertz). After the demultiplexer has separated the third of three voice samples from its current 16-bit voice data word, it requests another 16-bit voice data word from the FIFO memory. This process continues until the FIFO memory becomes empty or the voice decoder receives a stop command from the 7/32 processor.

Voice samples are retained in the voice sample latch, which in turn acts as the data source for the digital-to-analog converter. The output of the digital-to-analog converter is a voltage that is proportional to the data contained in the voice sample latch. Since the data in the voice sample latch change with each sample clock, a speech waveform is generated by the digital-to-analog converter.

The speech signal must be low-pass filtered at this time to remove some of the discontinuities which were created during the digitizing process. The filtered speech signal is then amplified and played-out via a speaker which is mounted on the message entry console.

MESSAGE ENTRY CONSOLE.

The message entry console is the position where a flight service specialist enters and reviews spoken messages into the Mass Weather Dissemination System. This position consists of a computer terminal for communication with the 7/32 processor for the selection of message segments to be updated and reviewed and the audio equipment required by the voice digitizer. Figure 7 is a functional diagram of the message entry console, and figure 8 is a picture of this position.

The computer terminal consists of standard off-the-shelf equipment including a Conrac RQB-14 cathode-ray tube (CRT) monitor, an Ann Arbor KB-300D keyboard, and an Ann Arbor K2480C CRT controller. The CRT controller interfaces the keyboard and CRT monitor to the 7/32 processor via an RS232 serial interface. The characteristics of this interface are: 4800 baud, 2 stop bits, 7 data bits, and no parity bit.

Data from the keyboard are sent to the 7/32 processor via the CRT controller as ASCII characters. Data from the 7/32 processor to be displayed on the CRT monitor are also sent as ASCII characters; however, the CRT controller first converts these ASCII characters into a video signal which is required by the Conrac monitor. The CRT controller can display the 64 uppercase characters of the ASCII character set, contains an internal refresh memory, and can display up to 24 lines of 80 characters each.

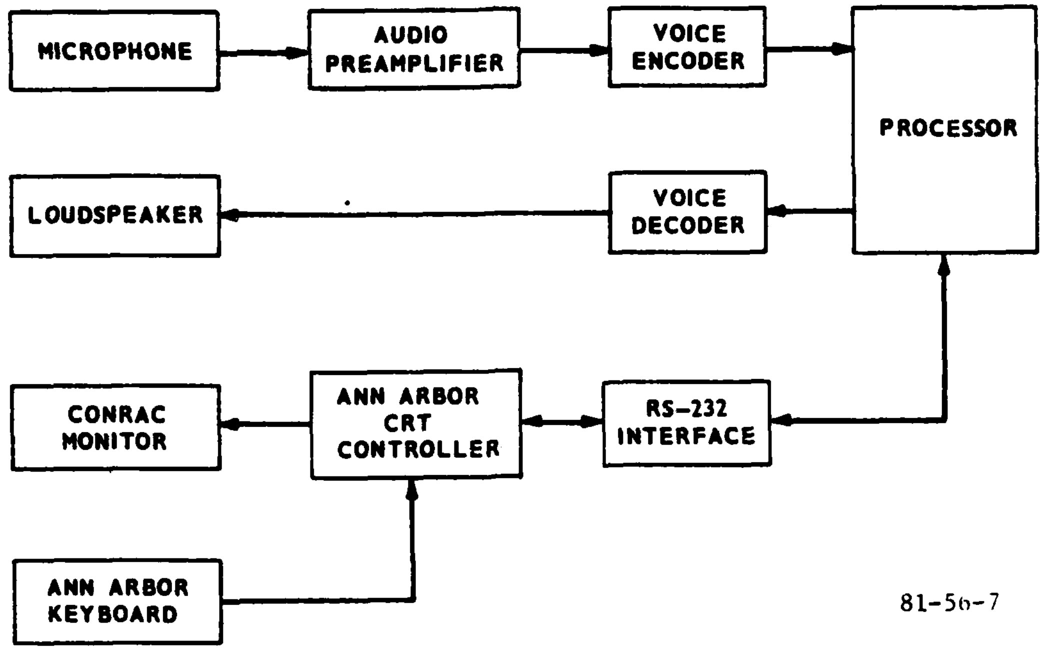


FIGURE 7. MESSAGE ENTRY CONSOLE

The audio equipment that is required by the voice digitizer consists of a loudspeaker and a microphone and preamplifier. The loudspeaker is connected to the voice decoder and allows the specialist to listen to previously entered spoken messages. The microphone and audio preamplifier provide the speech signal to the voice encoder which enables the specialist to enter spoken voice messages into the system.

The Technical Center's Mass Weather Dissemination System is configured to disseminate five different messages, each of which can be a maximum of 10 minutes duration. Each message is subdivided into 15 message segments. Messages are automatically composed by speaking message segments into the system one at a time. The duration of message segments can be variable in length, the only restriction being that the sum total time of all active segments cannot exceed the maximum system capacity of 87 minutes. Message segments which are common to all messages are updated simultaneously and automatically for all messages when a common message segment input is made.

The specialist controls the system through a CRT terminal. He enters commands via a keyboard and observes the reponse on the CRT. The commands provided for the specialist are entitled: TALK, REVIEW, UPDATE, KILL, CURRENT, MAP, LISTEN, PLAYBACK, DELETE, ABORT, and HELP.

FIGURE 8. SPECIALIST MESSAGE ENTRY CONSOLE

When the specialist types TALK on his console followed by a message segment name, the processor responds with a visual display of the word TALK, and the specialist then begins speaking the message segment into the system. The processor enables the voice encoder and begins storing digitized voice data into an appropriate voice data file. After he has spoken the message segment, the specialist depresses the carriage return key.

At this time, the message segment just spoken into the system is not yet available for dissemination. The UPDATE command must be executed before a message is available for dissemination.

The specialist can review the message segment just spoken into the system by typing the word REVIEW followed by the message segment name. The processor retrieves the appropriate voice data file, converts the digitized voice to audio in the voice decoder, and plays it to the specialist via a loudspeaker. When the specialist is satisfied with the message segment, he types UPDATE, followed by the message segment name. The UPDATE command replaces the current voice data file (if one exists) with the new voice data file. The old voice data file (if one exists) is put on a time-delayed delete list and will be deleted by the system automatically in 10 minutes. If the specialist is not satisfied with the previously spoken message segment, he can eliminate it by typing the word KILL. The KILL command deletes the current voice file immediately and makes its storage space available for the next spoken message segment. Should the specialist forget the name of the currently spoken message segment, he can retrieve it by typing CURRENT. The processor will respond with the name of the last message segment which was spoken into the system.

The MAP command allows the specialist to monitor a listing of the message segment content of any message. After typing MAP, followed by a message name, the processor sends the list of all possible message segment names and indicates which of the message segments are currently active (i.e., have data currently entered in the associated file).

The specialist can listen to any currently active message segment to verify its existence or voice quality. After typing LISTEN, followed by the message segment name, the processor retrieves the appropriate voice data file and plays it through the loudspeaker via the voice decoder.

An entire message can be monitored by the specialist by typing PLAYBACK and the message name. Each currently active message segment which constitutes the message is played through the loudspeaker via the voice decoder.

A message segment can be deleted by typing DELETE and the message segment name which is to be deleted. The voice data file for that message segment is then placed on a time-delayed delete list. Ten minutes later, that voice data file is automatically purged from the system.

The ABORT command allows the specialist to recover from an error. ABORT, followed by a message segment name, results in the current voice data file for that segment being placed on the time-delayed delete list. The old voice data file for that message segment becomes the current voice data file for that message segment. If the 10-minute timer had already elapsed for the old voice data file, an abort is not possible, an error message is sent to the CRT, and the current message segment remains unchanged.

By typing HELP, the specialist is presented with a list of the system commands on his CRT. Each command is followed by text which describes the command. Typing the HELP command displays the information as shown in table 1.

Extensive error correction is employed in the specialist command software. If either an illegal command, a legal command with an invalid message segment name, or a legal command which fails to meet appropriate prerequisites is attempted, the command is ignored, and an appropriate error message is sent to the CRT.

TABLE 1. LIST OF SYSTEM COMMANDS

Command	Operand	Definition
TALK	Segment Name	This command allows the specialist to prepare a message element.
REVIEW	Segment Name	This command is used to audition the current message element before entry into the system.
UPDATE	Segment Name	This command causes the current message element to be entered into the system with auto time-delay deletion of old segment, if one exists.
ABORT	Segment Name	This command cancels a previous update command by replacing the updated message segment with the old saved segment and deleting the update.
LISTEN	Segment Name	This command allows the specialist to hear any message segment in the system, if it exists.
DELETE	Segment Name	This command allows the specialist to delete any message segment in the system, if it exists.
MAP	Message	This command will map out the segment names of any one of the five messages and indicate which are active.
CURRENT	Update	This command allows the specialist to obtain the name of the present update segment, if one exists.
KILL	Update	This command allows the specialist to immediately delete the update before entry into the system.

BULK STORAGE SUBSYSTEM.

The bulk storage subsystem of the Mass Weather Dissemination System consists of two Perkin-Elmer moving head disc drives and their associated controller, five Amcomp fixed head disc drives and their C3 Incorporated controller, and a magnetic tape unit. Figure 9 shows the configuration of the bulk storage subsystem.

MOVING HEAD DISC. The moving head disc memory subsystem consists of a Perkin-Elmer M46-646 10-megabyte moving head disc storage system, with an M46-647 expansion drive. In this configuration, both 10-megabyte drives share a controller and a selector channel. Data transfers may be directed to or from only one drive at a time. Each drive has a storage capacity of 10 megabytes on two platters. One platter is fixed, and the other is a removable disc pack.

The average latency is 12.5 milliseconds with the worse case latency being 25 milliseconds. The seek time is 10 milliseconds for adjacent tracks, 40 milliseconds for a one-third of the disc seek, and 65 milliseconds for a full disc seek. Data are transferred at a rate of 312.5 kilobytes per second.

The first drive is used for the storage of the 7/32 operating system, system utility programs, operational and developmental programs, and data files. The second drive will be used for vocabulary storage by the Automatic Message Composition project.

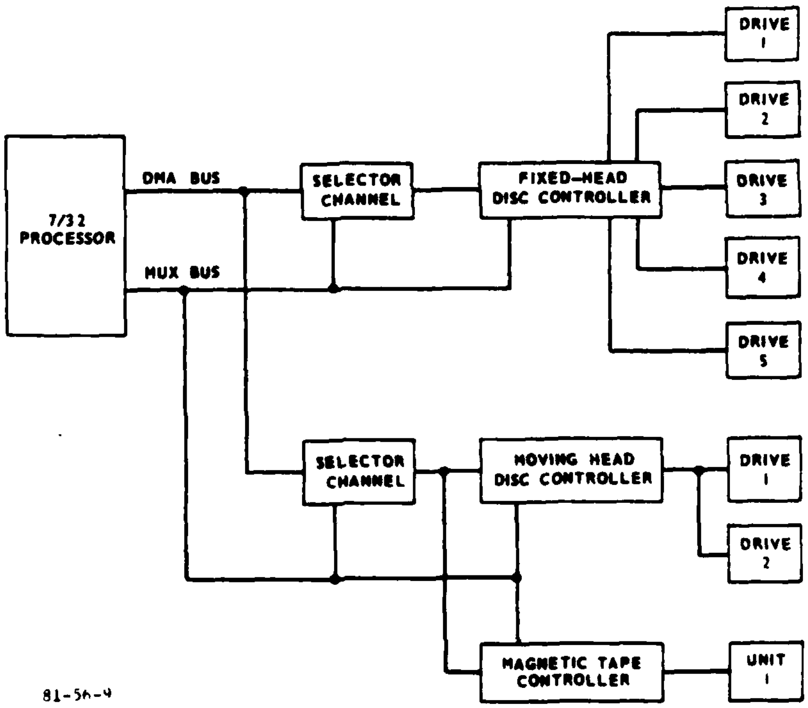


FIGURE 9. INTERDATA 7/32 BULK STORE SUBSYSTEM

FIXED HEAD-PER-TRACK DISCS. All of the digitized voice messages are stored in the fixed head-per-track disc bulk memory. The total memory capacity of the fixed head discs is 20 megabytes. This 20 megabytes of storage represents 87 minutes of spoken messages. This subsystem consists of five fixed head disc drives, a disc controller, and the software necessary to make the fixed head discs compatible with the 7/32 operating system. The fixed head disc bulk memory subsystem was supplied by C3 Incorporated; 11425 Isaac Newton Square South, Reston, Virginia 22090. C3 Incorporated is a software and hardware specialty company which specializes in Interdata (Perkin-Elmer) equipment.

The fixed head-per-track disc drives are Amcomp 8530 Disc Memory Units with a storage capacity of 4,194,304 bytes after formatting. The salient characteristics of these drives are a rotational speed of 3,600 rpm, an average access time of 8.33 milliseconds, a maximum access time of 16.66 milliseconds, and a data transfer rate of 9 million bits per second. Each drive has 256 tracks which consist of 256 64-byte sectors. The 8530 drive is highly modular, and repair is easily accomplished by swapping spare modules.

The disc controller is a proprietary device manufactured by C3 Inc. This controller is fabricated on an MDB Universal Logic Interface Module number MDB-48-013-16-01. Since the data transfer rate of this controller is 9 million bits per second, this controller must be connected to the 7/32 processor's highest priority selector channel, otherwise there is a possibility that data will be lost. Data buffering, timing, and error detection are accomplished by the controller. Data throughput is increased by this controller because it keeps track of each disc's position at all times.

The software driver supplied with the fixed head disc system makes the fixed head discs compatible with the 7/32 operating system. With this driver, all data requests are accomplished by using standard SVC's. An SVC is a supervisor call instruction that permits a user program (nonprivileged) running under operating system control to initiate certain privileged functions such as input/output.

MAGNETIC TAPE UNIT. A magnetic tape drive and its controller are also connected to the 7/32 processor via the same selector channel that is utilized by the moving head discs. The tape drive is an 800-BPI, 9-track drive which is used for making backups of the system programs and data files.

TELEPHONE VOICE OUTPUT CHANNEL.

The telephone voice decoder contains the logic required for the reconstruction of digitized voice samples back into an analog voice signal, as well as the logic required to handle the telephone control signals — ring, pickup, hangup, and customer connect. The telephone voice decoder interfaces to the telephone system via voice connecting arrangement CDH and connecting arrangement CBF. Figure 10 is a functional diagram of the telephone voice decoder. A detailed description of the telephone voice decoder is available in appendix C.

In addition to providing ring, pickup, hangup, and voice signals, this connecting arrangement provides a customer connect signal. This signal is active when a person is connected to the other end of the telephone line and is inactive when he hangs up. This signal is presented to the 7/32 processor as a status bit. As soon as the processor detects that the person on the other end of the line has hung up, it makes this line available for the next caller, thus maintaining telephone line utilization at a high level.

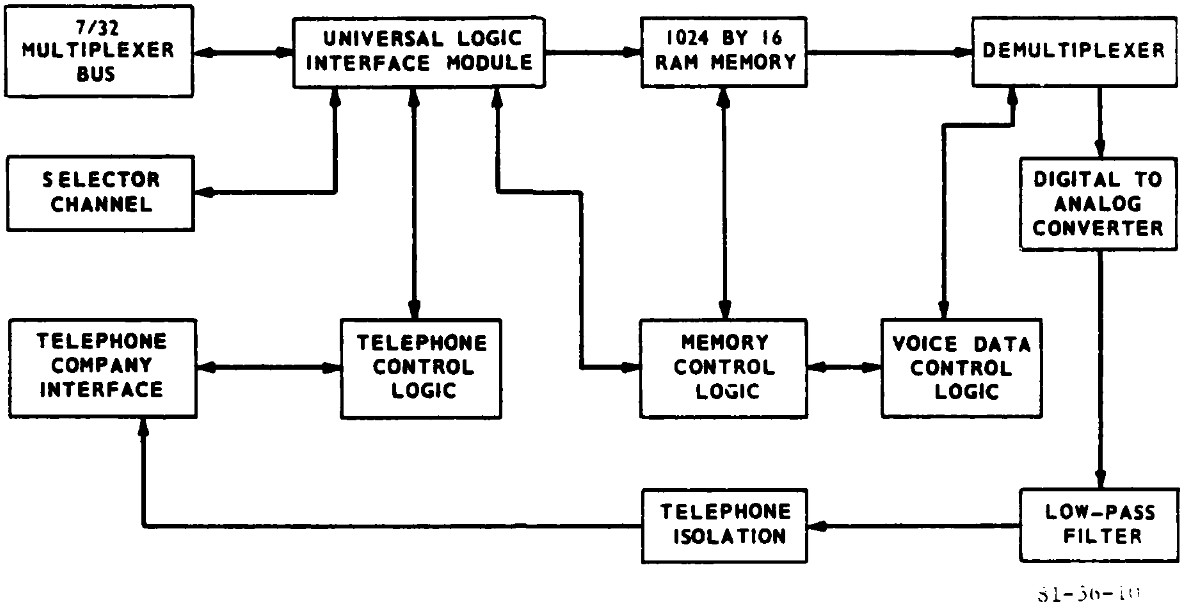


FIGURE 10. TELEPHONE VOICE DECODER

An operational sequence begins when the telephone voice decoder receives a ring signal from the telephone equipment. The ring signal causes the 7/32 processor to be interrupted via the Universal Logic Interface Module's (ULIM) control logic. The processor detects that it has received a ring interrupt and responds by sending a pickup command and a start command.

The pickup command causes ring signals to terminate and enables the telephone line for the transmission and receipt of voice signals. The start command enables the telephone voice decoder for operation and prepares it for the receipt of digitized voice data.

Voice data are immediately sent to the telephone voice decoder via the processor's selector channel. This voice data go directly into the telephone voice decoder's internal memory. The internal memory is included so that the telephone voice decoder is not constantly interrupting the 7/32 processor for data.

The internal memory is divided into two equal parts of 512 16-bit words each. With this quantity of memory, each active telephone voice decoder module interrupts the 7/32 processor every 0.25 seconds for more data. The internal memory is included to reduce the number of service requests for data to the 7/32 processor, which in turn reduces system overhead. For purposes of discussion, assume the memory halves are labeled A and B.

While digital voice data are being taken from A and being converted to audio, the processor is filling B with the next segment of digital voice data. When the last data word has been taken from A, a switch is made such that the next data will be taken from B. While the data in B are being converted to audio, the processor is filling A with the next segment of digital voice data. When the last data word has been taken from B, a switch is made such that the next data will be taken from A. This process continues until an entire voice message has been completed.

The memory control logic is responsible for keeping track of the memory halves as previously described. It constantly keeps track of the exact memory location of data to be converted to speech next, as well as the exact memory location to be updated by the processor next. This logic also prevents problems from occurring when the processor and voice data control request the memory simultaneously. The memory control logic delays a memory request by the processor, while the voice data control is retrieving a data word from memory, and delays the voice data control from reading a data word from memory, while the processor is updating a memory location. When the voice data control has emptied its half of the memory, the memory control interrupts the processor for the next batch of digital voice data.

Data from the memory are latched in a voice data register. These data are connected to a demultiplexer which separates the three 5-bit voice samples from the 16-bit word. When a 16-bit word has been completely disassembled, another 16-bit word is requested from the memory via the voice data control logic. This process continues until the memory is empty.

Five-bit voice samples are applied to the inputs of a companding digital-to-analog converter. Here, the digital voice samples are converted back into a speech signal.

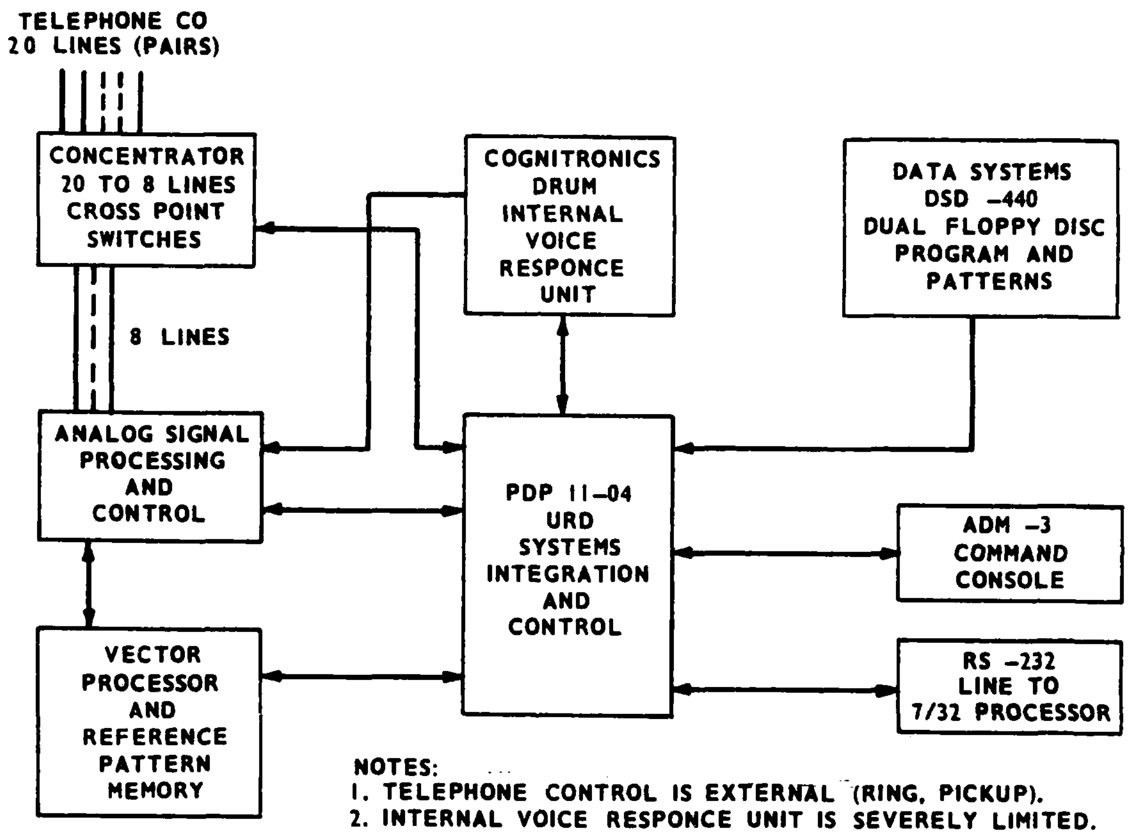
The raw speech signal from the digital-to-analog converter contains sharp discontinuities as a result of the quantizing (analog-to-digital) process. These are removed by passing the speech signal through a low-pass filter.

The last stage of the filter drives a 600-ohm transformer. This transformer provides the isolation required by the telephone company, as well as impedance matching.

There is also a switch between the transformer and the telephone equipment. When the memory has no more data to be converted into speech, the memory control opens the switch, thereby disconnecting the loading effects of the telephone voice decoder on the telephone line when the telephone voice decoder is not producing speech. This improves the performance of other system components, such as the fast-file recorder and the utterance recognition device.

UTTERANCE RECOGNITION DEVICE.

The URD is a Verbex (formerly Dialog Systems, Inc.) Model 1800 Speech Recognition Terminal. A block diagram of the URD is shown in figure 11. A picture of the URD is shown in figure 12. The URD's role in the Mass Weather Dissemination System is to recognize pilot's responses to questions. The URD's decision as to which word a pilot has spoken is based on statistics which are based upon how many people have said the word. A caller using the system must use the correct set of words just as he must use the proper telephone number to access the system. The pilot's response is coded and sent to the 7/32 processor which takes the appropriate action. Thus, the pilot controls the Mass Weather Dissemination System and custom tailors the briefing to suit his needs.



NOTES:

1. TELEPHONE CONTROL IS EXTERNAL (RING, PICKUP).
2. INTERNAL VOICE RESPONCE UNIT IS SEVERELY LIMITED. VOCABULARY WORDS PLUS SHORT PROMPTS SUCH AS "PLEASE REPEAT" AND "WAS THAT"
3. SYSTEM CONTROL IS EXTERNAL. HOST PROCESSOR SPECIFIES URD CHANNEL NUMBER AND TELEPHONE LINE NUMBER. URD "LISTENS" TO THE SPECIFIED LINE AND SENDS A RESPONSE TO THE HOST PROCESSOR.

81-56-11

FIGURE 11. FAA TECHNICAL CENTER URD

FIGURE 12. UTTERANCE RECOGNITION DEVICE

The URD is a discrete word recognition machine; that is, it is capable of recognizing only a single word at a time from a preprogrammed vocabulary. The URD may be instructed to look for words within the entire vocabulary or it may be instructed to look for words within logical subgroups. The smaller the subgroup, the higher the recognition rate. Since the recognition rate of the URD is best when considering only a portion of its vocabulary, recognition by subgroup is performed wherever possible within the Mass Weather Dissemination System. The complete set of vocabulary words that the URD can recognize is:

Affirm/Deny Words:	Affirmative, Negative, Yes, No
System Option Words:	Briefing, File, Specialist, Amend, Close
Briefing Direction Words:	North, South, East, West, Local
Numbers:	One, Two, Three, Four, Five, Six, Seven, Eight, Nine, Zero, Niner

The URD is very easy to operate. Depression of the reset switch on the front panel of the PDP 11-04 causes an autoload sequence that loads the operational program and vocabulary reference patterns. A message is sent to the ADM-3 terminal which indicates the URD's status.

The interface between the URD and the 7/32 processor is serial RS-232 interface. The serial interface between the machines has the double advantage of low cost and the elimination of the need to write a special software driver. Command messages from the 7/32 and response messages from the URD are accomplished by the transmission of ASCII characters.

The URD is a multichannel device which can listen to eight speakers simultaneously. The URD also has a self-contained switching array which can switch any of its 8 input channels to any of the 20 incoming telephone lines. The 7/32 processor controls this switching via commands over the serial interface.

The URD also contains an internal voice response unit. This unit allows the URD to make corrections when it is unsure what the speaker said by responding with "Please repeat" or "Was that XXX?", where XXX is one of the vocabulary words.

The audio inputs to the URD are switched telecommunications lines. This implies that the audio level is different for every connection and that the high and low frequency components of the spoken words are missing. Most comparable voice recognition machines require "clean audio input."

The URD is an untrained device; that is, anyone can use this machine with an equal probability of success. Most comparable voice recognition machines are trained to the person who is using it. The specifications for the URD were that it would recognize 95 percent of the people who used it. We found that the URD met or exceeded this specification when recognition by subgroups was employed.

TELEPHONE TERMINAL EQUIPMENT.

The Mass Weather Dissemination System has, as its interface to the telephone system, special telephone terminal equipment which was installed in the laboratory. Telephone company nomenclature for this equipment is voice connecting arrangement CDH and connect arrangement CBF.

This connecting arrangement was chosen because of the availability of a signal which indicates whether a pilot is actually connected to the line. Thus, the signal would indicate that a pilot has hung up in the middle of a briefing, so that the system could make that line available for the next caller.

The audio lines are connected to the output channel interface of the 7/32 computer, the crosspoint switch and the URD. These audio lines are connected in such a manner to allow a two-way flow of speech information between the pilot and the system.

The control signals — ring, pickup, and customer detect — are connected between the telephone interface equipment and the output channel interface of the 7/32 computer. There is one interface for each telephone line serviced.

When a pilot calls the system, the ring signal goes true. The computer "picks up" the line by making and holding a relay connection between the pickup control lines. To "hang up" the telephone, the computer merely breaks this connection. The customer detect signal was previously discussed.

The telephone company provided a "ring-down" feature for the 20 telephone lines. This feature enabled the pilot to dial only the first number in the series of telephone numbers assigned to the Mass Weather Dissemination System. The telephone company equipment automatically connected the pilot to the next available telephone line in the series of telephone lines.

TELEPHONE LINE CONTROLLER.

All of the telephone line switching within the Mass Weather Dissemination System is performed by the crosspoint switch. The crosspoint switch can connect any of the 20 incoming telephone lines to either of the two fast-file flight plan recorders or to any one of the four flight service specialist positions. Control of the crosspoint switch is achieved via data and commands which are received from the 7/32 processor. The crosspoint switch subsystem is shown in figure 13. A more detailed description of the crosspoint switch is contained in appendix D.

The heart of the crosspoint switch is a Clare 969-A48-A2E Mini Memory Matrix Module. Each of these modules is a self-latching, 8 by 8 crosspoint switch. Control of this module is provided via 16 data bits and a strobe pulse. Eight of the data bits represent eight of the incoming telephone lines, while the remaining eight data lines represent system options, such as fast-file flight plan recorders and specialist positions. After the 16 data bits have been established, a command generates a strobe pulse which actually performs the switching function. The module holds this connection until instructed to change it.

Three Clare modules were interconnected to form a 24 by 8 crosspoint switch. Of this capacity, 20 x 6 was actually utilized by the Mass Weather Dissemination System.

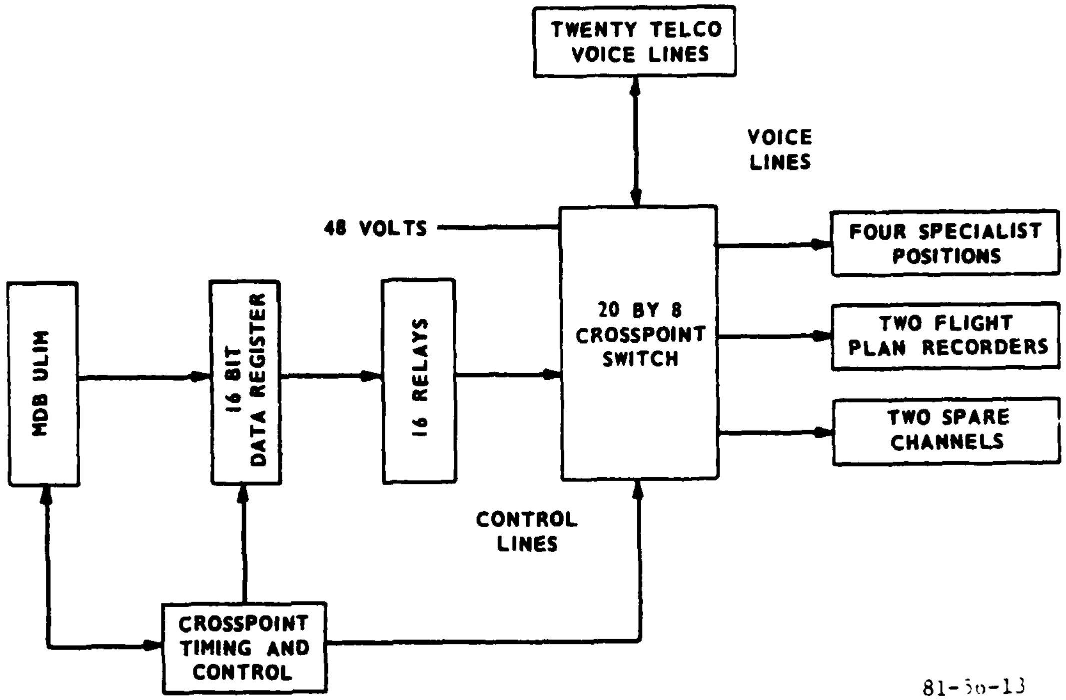


FIGURE 13. CROSSPOINT SWITCH SUBSYSTEM

The crosspoint switch is utilized when the 7/32 processor determines that a pilot on one of the incoming telephone lines has requested to be connected to a system option. The 7/32 determines which of the system options is available for use so that it can generate the proper 16-bit control word, as well as the proper command. The 7/32 processor then sends the 16-bit control word followed by the command to the crosspoint switch controller, and the switching is performed. The pilot is now connected to the system option of his choice.

SPECIALIST ANSWERING SET.

The Mass Weather Dissemination System incorporates four specialist positions. These positions consist of four standard telephone instruments which were modified to be compatible with the Mass Weather Dissemination System. The specialist telephone instruments can be connected to any of the 20 incoming telephone lines and are used when a pilot requests to speak to a flight service specialist, or when the URD fails to understand a pilot's request and a specialist must assist the pilot to get the information that the pilot requires.

Figure 14 is a block diagram of one of the specialist positions. Figure 15 is a picture of a specialist position. A detailed description of the specialist position is included in appendix E.

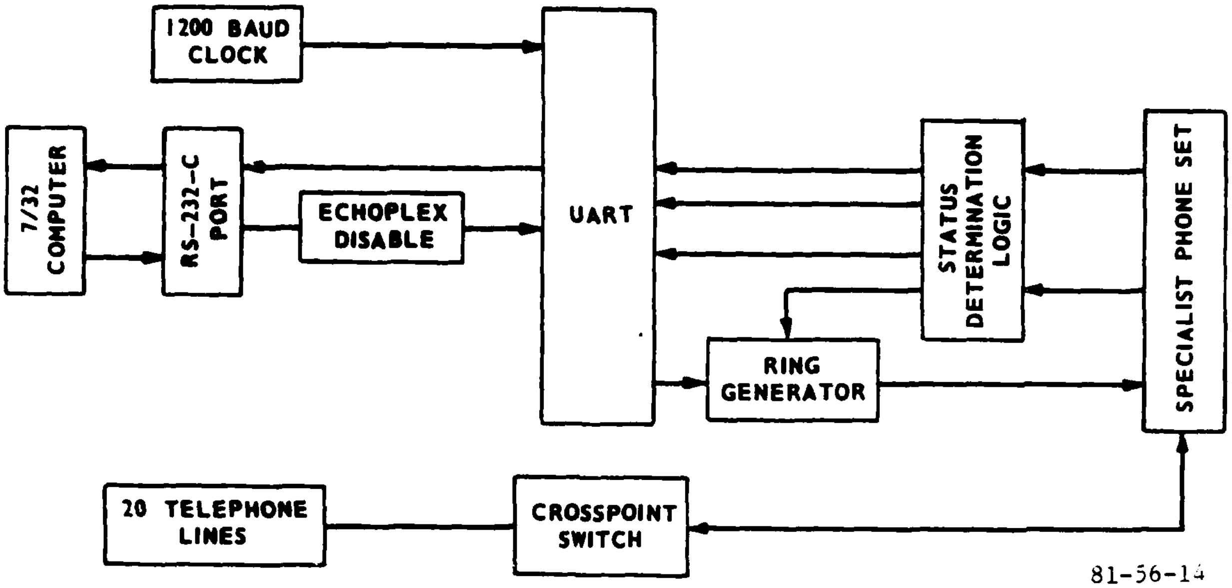


FIGURE 14. SPECIALIST POSITION

Audio connection of the pilot on one of the incoming telephone lines is accomplished via the crosspoint switch, which is described in this report. Control of the specialist position is achieved via an RS-232 asynchronous serial line interface.

The use of the serial interface enabled the system to use a cheap serial interface to control the specialist positions and eliminated the need to write a special software driver. Since the system serial interface software driver utilized echoplex, a circuit in the specialist phone control logic disabled the echoplex feature.

A General Instruments AY-5-1013 Universal Asynchronous Receiver Transmitter (UART) is the key device for transmitting data between the 7/32 computer and the specialist telephone instrument. By sensing switches on the telephone instrument, the computer could determine if the phone was on-line or off-line and whether the instrument was picked up or not. A 1200 baud clock provides all of the timing signals required by the specialist position.

If a specialist had to leave his position to perform another duty, he would put his phone in the off-line state by flipping a switch. The 7/32 computer would know that it could not connect a pilot to this particular position, so it would search for the next available specialist phone.

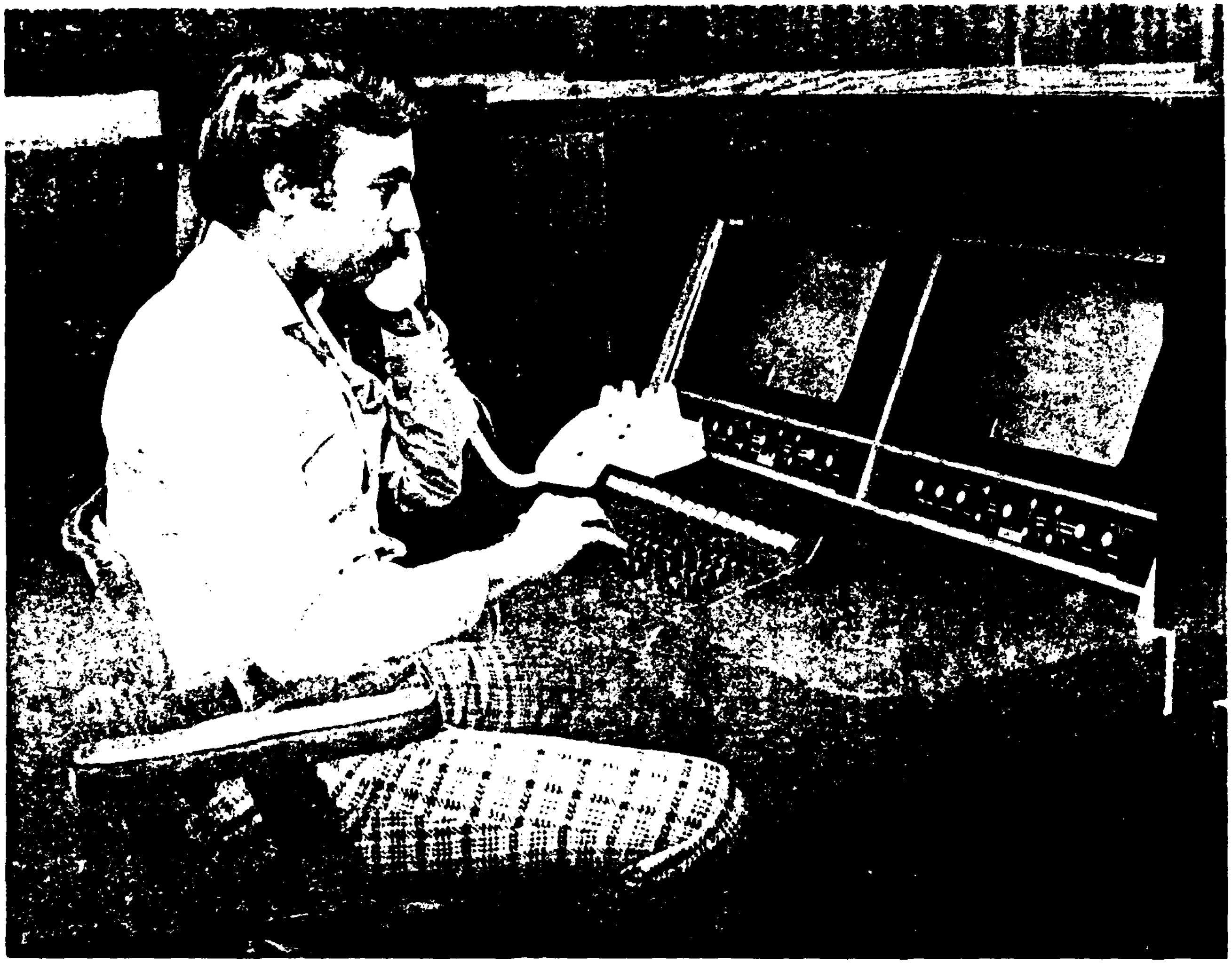


FIGURE 15.   SPECIALIST ANSWERING SET AT CONSOLE

The computer can send only one command to the specialist position — the ring command. This causes a buzzer to buzz and a light to flash. The buzzer can be disabled by flipping a switch.

FAST-FILE RECORDER.

The two fast-file recorders associated with the Mass Weather Dissemination System are used for the filing, closing, and amending of flight plans. Either of the fast-file recorders may be connected to any of the 20 incoming telephone lines by means of the crosspoint switch. These recorders differ significantly from present flight plan recorders in that the pilot is given the option of reviewing his flight plan and amending it, if required.

The fast-file recorders are modified Lanier Tele-Edisette 1400C dictation units which are equipped with voice operated relay circuits. The modifications are primarily concerned with the conversion from manual operation to automatic remote control operation by the 7/32 computer.

Figure 16 is a block diagram of a fast-file recorder. Figure 17 is a picture of the fast-file recorders and the flight plan entry console. A detailed description of the fast-file recorder is included in appendix F.

Control of the fast-file recorder is achieved via an RS-232 asynchronous serial line interface. The use of the serial interface enabled the system to use a cheap serial interface to control the fast-file recorders and eliminated the need to write a special software driver. Since the system serial interface software driver utilizes echoplex, a circuit in the fast-file recorder control logic disables the echoplex feature.

A General Instruments AY-5-1013 UART is the key device for transmitting control and status data between the 7/32 computer and the fast-file recorder. A 1200 baud clock provides all of the timing signals required by the AY-5-1013.

After the 7/32 processor has determined that a pilot wants to file a flight plan, it plays out the instructions on how to use the fast-file recorder to the pilot and then sends the sense status command to the fast-file recorder. If the status received from the recorder is bad, it goes to the next recorder and checks its status. If the status is bad, the pilot is put on a wait list until a recorder becomes available.

When a recorder becomes available, the 7/32 computer sends a start command. The fast-file recorder sends a tone to the pilot and begins to record his flight plan. When the pilot has finished entering his flight plan, the voice activated relay informs the 7/32 computer that recording has been terminated. The computer then asks the pilot if he would like to review his flight plan.

If the pilot says "yes," the computer rewinds the tape and plays the flight plan back to him. When this is completed, the computer asks the pilot if he would like to make amendments to his flight plan. If the pilot says "yes," the computer starts up the fast-file recorder once again. When the pilot finishes entering his amendments, he remains silent and the computer asks him if he would like to listen to his flight plan amendments. If he says "yes," his amendments are played back and he is given another chance to make amendments. This process continues until the pilot has no further amendments.

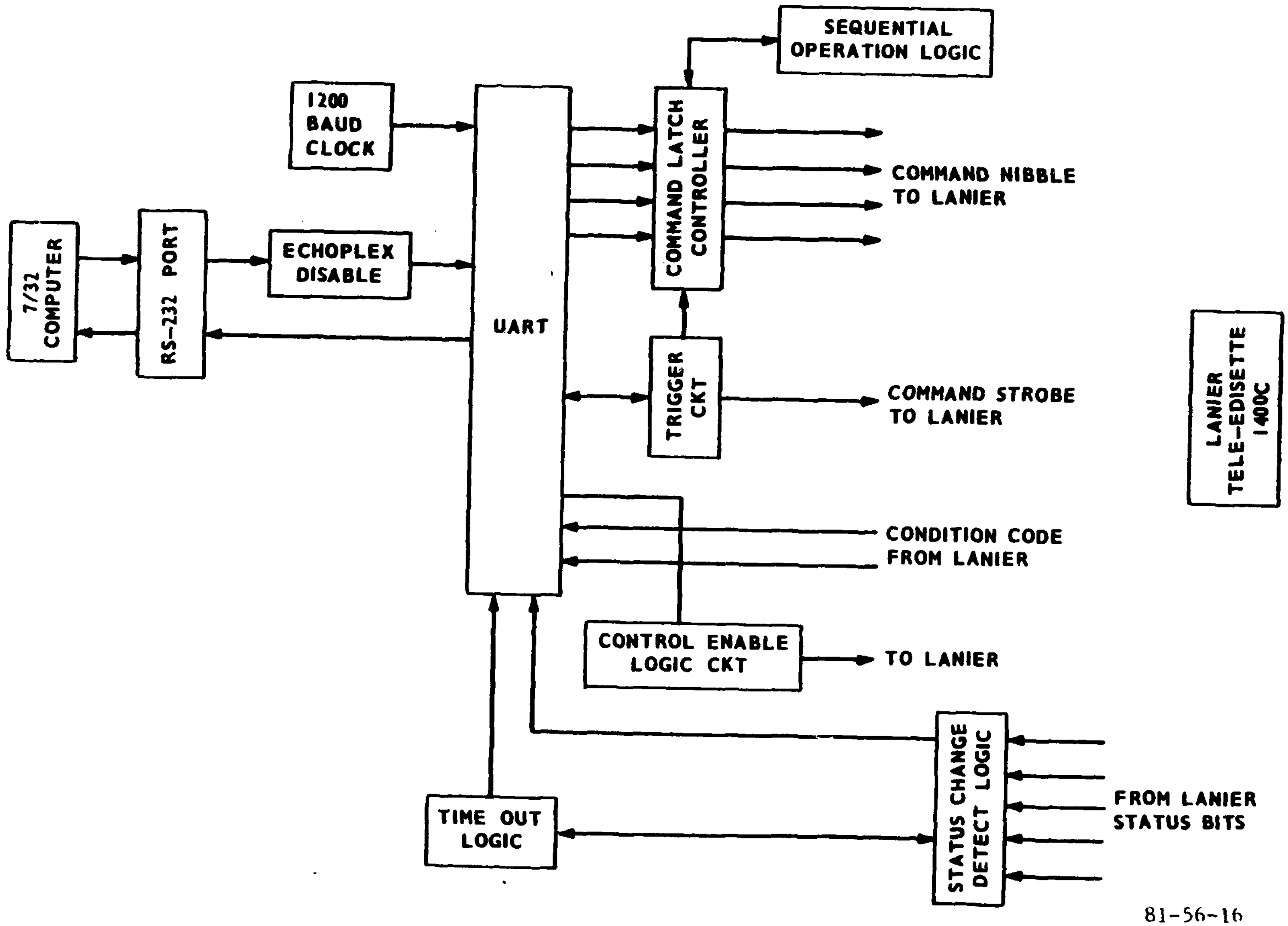


FIGURE 16. FAST-FILE SUBSYSTEM

FIGURE 17.   FAST-FILE RECORDERS AND FLIGHT PLAN ENTRY CONSOLE

If the pilot says "no" after being asked to review his flight plan or says "no" after being asked if he wants to make amendments to his flight plan, the computer asks him if he requires further assistance. If he says "no," the computer says "Thank you, have a good day" and hangs up. If he says "yes," he is given the opportunity to select any of the system options — briefing, file, specialist, amend, or close, once again.

FLIGHT PLAN ENTRY CONSOLE.

The flight plan entry console is the position where information recorded on the fast-file flight plan recorders is transcribed into flight plan form. Flight plans are entered and edited via a CRT display and keyboard which are interactive with the 7/32 processor. Automatic formatting of the flight plan for transmission over service B is provided. As an option, flight plans may be printed locally.

The CRT display is a Conrac model QQA-14/N. The keyboard is an Ann Arbor model KB 300D. Interface to the 7/32 processor is accomplished via an Ann Arbor K2480C controller.

PRINTER.

The printer allows for the hard copy output of system parameters and flight plans. It can also serve as a backup system console if the need arises. Program listings were generated using the printer during system development.

The printer is a Perkin-Elmer M46-010 Carousel printer. This device prints the uppercase ASCII characters at a rate of 30 per second. Interface to the 7/32 processor is accomplished via an RS-232 20-milliampere current loop.

AUTOMATIC MESSAGE ENTRY CONSOLE.

The automatic message entry console is the position where textual messages are generated for use by the Automatic Message Composition System. Plans originally called for this console to be remotely located at a National Weather Service Office at a later time in the Automatic Message Composition System's development. The existing console is an ADM-3 CRT terminal that demonstrates how the position would work by operating with the existing test vocabulary of 250 words. Figure 18 is a picture of the proposed automatic message composition entry terminal which was to have operated at a National Weather Service Office.

SYSTEM OPERATION

INTRODUCTION.

The pilot initiates the transaction by dialing the appropriate telephone number to gain access to the system. Upon receipt of the third ring, the system picks up the line and says "Hello ...This is the Pilots' Automated Briefing System, detailed operating instructions are available for the new or occasional user. Do you wish detailed operating instructions? Please say yes or no immediately after the cue tone."

FIGURE 18. AUTOMATIC MESSAGE COMPOSITION ENTRY TERMINAL

If the pilot answers "yes" to this question, the machine responds "During the briefing, you will be asked to say certain words to the computer. We ask that you speak clearly, only one word at a time, and only immediately after the cue tone. If the computer asks you to Please repeat, say your last word again immediately after the cue tone. If the computer questions your last word, for example, 'Was that north?' Please say the word Yes or No immediately after the cue tone. If the computer does not respond in approximately 3 seconds, please say your last word again, louder. Do you wish a replay of the information just presented? Say yes or no immediately after the cue tone."

If the pilot answers this question by saying "yes," the instructions will be repeated and will continue to be repeated until he eventually says "no." At this point, he proceeds to the options question as does the pilot who answered no to the first question. The ONLY difference will be the detail of the system operating instructions. The pilot who responds to the first question by saying "no" will receive brief operating instructions which will be shown in parenthesis for the remainder of this section of the report. Regardless as to whether brief or detailed operating instructions are requested, the pilot's options within the system are identical.

The system now responds with the options question: "You may now select one of five system options. You may select any of four route-oriented briefings or a local area briefing. You may speak your flight plan into fast-file, amend a previously entered flight plan, or close out a flight plan. You may speak to a flight service specialist. You may terminate this call at any time by merely hanging up. Immediately after the cue tone, please say briefing, file, specialist, amend, or close." (The short form of this statement is "Say briefing, file, amend, specialist or close.")

The pilot will now select one of these five options. The briefing option will be discussed first, followed by the specialist, file, amend, and close options. At the completion of all of these options, the pilot is asked "Do you require any further information or assistance?" (The short form of this question is "Do you require additional assistance?") If the pilot answers yes, he is once again presented with the options question. If he says "no," the computer says "Thank you, have a good day" and then hangs up.

BRIEFING OPTION SELECTED.

If the pilot answers the options question with the word briefing, the computer says "To select a generally northbound route-oriented briefing, say the word north. Similarly say east, south or west for other route-oriented briefings or the word local for a local area briefing. Immediately after the cue tone, say north, east, south, west or local." (The short form of this statement is "Say north, south, east, west, or local.") The pilot waits for the beep tone and then says "north," for example. The computer gets the north message from the bulk storage and begins to play it to the pilot. When the computer finishes playing the north message to the pilot, it asks "Do you require additional assistance?" If the pilot responds "no", the computer says "Thank you, have a good day" and then hangs up. If the pilot responds "yes," the computer responds with the options question as was previously discussed. All of the other briefing choices — south, east, west, and local — work the same way.

SPECIALIST OPTION SELECTED.

When the pilot answers the options question by saying "specialist," the computer responds "Stand by for a specialist." The abbreviated form of this question is identical. At this time, the computer connects the pilot to the next available specialist position. After the pilot has received the information that he required from the flight service specialist, the specialist hangs up his phone. The computer then says to the pilot "Do you require additional assistance?" If the pilot answers "no," the computer says "Thank you, have a good day" and then hangs up. If he says "yes," the computer responds with the options question as was previously discussed.

FILE OPTION SELECTED.

The pilot may respond to the options question by saying "file." The computer will respond "You will be connected to the fast-file system, stand by. Immediately after the cue tone, speak your flight plan. The computer will interpret 5 seconds of silence as the completion of your flight plan. When you have finished, remain silent until the computer comes back to you." (The short form of this question is "Stand by to speak your flight plan immediately after the cue tone.") At this time, the computer switches the pilot to the next available fast-file recorder. The fast-file recorder issues a tone when it is ready to record, and the pilot speaks his flight plan. When he has finished entering his flight plan, the pilot remains silent and the computer says "Do you wish to play back your flight plan? Please say yes or no." (The short form of this question is "Do you wish your flight plan played back?")

If the pilot says "no," the computer asks "Do you require additional assistance?" If the pilot answers "no," the computer says "Thank you, have a good day" and then hangs up. If he says "yes," the computer responds with the options question as was previously discussed.

If the pilot wants to verify his flight plan, he says "yes." The computer then rewinds the fast-file recorder to the beginning of the pilot's flight plan and begins playing it back to him. At the completion of playing back the flight plan, the computer asks "Do you wish to make amendments to your flight plan? Please say yes or no." (The short form of this question is "Do you wish to amend your flight plan?")

If the pilot does not wish to make amendments to his flight plan, he says "no" and the computer then asks "Do you require additional assistance?" If the pilot responds "no", the computer says "Thank you, have a good day" and then hangs up. If he says "yes," the computer responds with the options question as was previously discussed.

If the pilot would like to amend his flight plan, he says "yes" and the computer responds "You will be connected to the fast-file system, stand by. Immediately after the cue tone, speak your flight plan amendments. Say the word amend, identify yourself, your aircraft, and previously entered estimated time of departure. When you have finished, stand by, the computer will come back to you." (The short form of this statement is "Stand by to speak your flight plan amendments immediately after the cue tone.") The fast-file recorder issues a cue tone and the pilot then enters his flight plan amendments.

At the completion of the pilot's entry of his flight plan amendments, the computer asks "Do you wish to play back your flight plan amendments? Please say yes or no." (The short form of this question is "Do you wish to play back your flight plan amendments?") If the pilot answers "no," the computer responds "Do you require additional assistance?" A "no" response results in the computer response "Thank you, have a good day" and then the computer hangs up. A "yes" response results with the computer giving the options question as was previously discussed.

If the pilot wants his flight plan amendments played back, he says "yes." The computer rewinds the fast-file recorder to the beginning of the pilot's flight plan amendments and begins to play them back to him. After completing the playback of the pilot's flight plan amendments, the computer asks "Do you wish to amend your flight plan?" The pilot is given yet another chance to make amendments to his flight plan. He is given the opportunity to continue to make amendments to his flight plan until he eventually answers this question by saying "no."

AMEND OPTION SELECTED.

The pilot may respond to the options question by saying "amend." The computer will respond "You will be connected to the fast-file system, stand by. Immediately after the cue tone, speak your flight plan amendments. Say the word amend, identify yourself, your aircraft, and previously entered estimated time of departure. When you have finished, stand by, the computer will come back to you." (The short form of this statement is "Stand by to speak your flight plan amendments immediately after the cue tone.") The fast-file recorder issues a cue tone and the pilot then enters his flight plan amendments.

At the completion of the pilot's entry of his flight plan amendments, the computer asks "Do you wish to play back your flight plan amendments? Please say yes or no." A "no" response to this question results in the computer response "Do you require additional assistance?" If this question is answered "no," the computer says "Thank you, have a good day" and then hangs up. A "yes" response to this question results with the computer asking the options question which was previously discussed.

If the pilot wants to verify his flight plan amendments, he says "yes." The computer responds by rewinding the fast-file recorder to the beginning of the pilot's flight plan amendments and begins to play them back to him. After completing the play back of the pilot's flight plan amendments, the computer asks "Do you wish to amend your flight plan?" The pilot is given yet another chance to make amendments to his flight plan. He may continue to make amendments to his flight plan until he eventually answers this question by saying "no."

CLOSE OPTION SELECTED.

The final way that the pilot could answer the options question would be to say "close." The computer would respond to this word by saying "You will be connected to the fast-file system, stand by. Immediately after the cue tone say the word close followed by your name, aircraft ID, and the time of day." (The short form of this statement is "Stand by to close out your flight plan immediately after the cue tone.") The pilot then closes out his flight plan by speaking this information into the fast-file recorder.

After the pilot enters his flight plan closure, the system asks "Do you require additional assistance?" If the pilot says "no," the computer says "Thank you, have a good day" and then hangs up. If the pilot says "yes," the computer responds by asking the options question which was previously discussed.

VOICE RECOGNITION ERRORS.

Since the Mass Weather Dissemination System uses a speaker independent voice recognition device over standard grade telephone lines, there is a good chance that voice recognition errors will occur. Some of the errors are caused by telephone system equipment, some are caused by the URD, and others are caused by the pilot himself.

Telephone system equipment errors are caused by poor connections, burst noise, and limited bandwidth. Poor connections result with a very low level of audio signal being presented to the URD. Sometimes when the audio level is good, the audio signal and noise are of equal levels. Burst noise is random noise on an otherwise good telephone connection, such as occasional tones and clicks. Restricted bandwidth prevents the URD from receiving all of the information that is contained in speech, the S and F sounds, for example.

The pilot is responsible for most of the voice recognition errors. If he does not obey the rules, he cannot play the game. If he answers the options question by saying "I want a northbound briefing," the URD will definitely have problems understanding him since it is only looking for a one-word response. If the pilot says a valid word or a nonvalid word before or during a cue tone, the URD will fail to recognize him. This was the largest single cause of voice recognition errors. It is imperative that the pilot speaks only after the cue tone! Even if he obeys the rules, the URD may not recognize him because he subconsciously slurs his words, he speaks a dialet that the URD does not understand, or he has a speech impediment.

The URD is responsible for the remainder of the voice recognition errors. The URD is an imperfect device that is constantly being updated as advances are made in the field of voice recognition. To a great extent, the voice recognition rate of the URD is dependent upon its stored vocabulary patterns. The set of vocabulary patterns currently within the URD is definitely not a "universal set." It remains to be seen if such a "universal set" of reference patterns is even attainable.

Some of these voice recognition errors are minimized, but not eliminated, by restricting the URD to consider only a small part of its entire vocabulary at any given time. For example, after the computer asks the options question, it instructs the URD to listen for only the words file, specialist, briefing, amend and close. All of the other words in the URD's vocabulary are considered to be nonvalid at this time.

The system uses two methods for error detection and correction of voice inputs. One of the methods is done at the system level where the computer tells the pilot that it is having problems understanding him and it gives him the opportunity to say the word again. The second method is accomplished within the URD, which performs internal error checks that will be discussed next.

URD INTERNAL ERROR CORRECTION.

The process is started when the 7/32 computer instructs the URD to listen for a word. After the pilot says the word, the URD makes a list of key parameters of his spoken word. The URD then compares the key parameters of the spoken word with sets of parameters of words in its vocabulary. Thus, a score is generated comparing the spoken word with every word in the URD's vocabulary.

The URD now examines the set of scores. If the best score is within the good range and is separated from the second-best score by at least a minimum amount, the URD considers itself to have recognized a word. The code for the recognized word is then sent to the 7/32 computer.

When the best score is within the good range but is not separated from the second-best score by a minimum amount, the URD is not sure that it has recognized a word. The URD then asks "Was that XXX?", where XXX is the first choice word. If the pilot responds "yes," the code for the first choice word is sent to the 7/32 computer. If the pilot says "no," the URD sends a not-recognized code to the 7/32 computer.

When the best score is within the marginal range, the URD is not sure that it has recognized a word and it responds to this situation by asking "Was that XXX?", where XXX is the word that had the best score. The yes or no sequence was previously described.

When the best score is within the bad range, the URD knows that it has not recognized a word and it responds by saying "Please repeat," thus giving the pilot another opportunity to say the word. If the best score is once again within the bad range, the URD sends a not-recognized code to the 7/32 computer. If the score is not within the bad range, the sequence of events described in the previous three paragraphs occurs.

SYSTEM ERROR CORRECTION.

Basic error detection and correction are handled within the URD. When the URD detects a problem, it sends a not-recognized code to the 7/32. The 7/32 computer then makes a second attempt to get the information that it needs from the pilot. The 7/32 computer does this by telling the pilot the list of valid words, once again, followed by a warning to speak only after the cue tone.

When the 7/32 computer receives a not-recognized code from the URD in response to the options question it responds "The computer did not understand you. Please speak clearly immediately after the cue tone, one of the following words: briefing, file, amend, specialist, or close." (The short form of this statement is "The computer did not understand you. Say again, briefing, file, amend, specialist or close.")

A valid response from the URD results in the pilot receiving the option that he requested as was previously discussed. A not-recognized code causes the computer to say "The computer did not understand you. Please stand by for a specialist." At this point, the specialist can assist the pilot with his briefing.

If the computer receives a no-response code as an answer to the route selection question, it says "The computer did not understand you. Please speak clearly immediately after the cue tone only one of the following words: north, south, east, west or local." (The short form of this statement is "The computer did not understand you. Say again, north, south, east, west or local.") A valid response to this statement results with one of the briefings being played to the pilot. A no-response code forces the computer to say, "The computer did not understand you. Please stand by for a specialist." At this point, the specialist can assist the pilot with his briefing.

In response to a not-recognized code from the URD for a yes or no type question, the computer says "The computer did not understand you. Please speak clearly after the cue tone either the word yes or no." (The short form of this statement is "The computer did not understand you. Say again yes or no.") A valid response to this question allows the pilot to continue with his briefing in a normal manner. When the computer receives a not-recognized code as an answer, it says "The computer did not understand you. Please stand by for a specialist." At this time, a flight service specialist will assist the pilot with his briefing.

The computer can also detect that a fast-file recorder is jammed. If the fast-file recorder is jammed and the pilot responds to the options question by saying file, amend, or close, the computer says "The fast-file recorder has jammed. Stand by for a flight service specialist who will accept your flight plan. Please advise the specialist that the recorder is jammed." The specialist can now take flight plan information from the pilot. The pilot is returned to the system when the specialist hangs up his telephone.

CONCLUSIONS

The design, development, and fabrication of the Mass Weather Dissemination System Exploratory Engineering Model led to the following major conclusions:

1. Synchronous access of messages was possible for multiple users selecting any mix of available messages.

2. Message-to-line multiplexing was accomplished by using an Utterance Recognition Device (URD) and a processor to take appropriate action based on the URD's responses.

3. The total message content of the system of five messages with a maximum duration of 10 minutes each was easily accomplished, since the total recording time of the system was 87 minutes.

4. One telephone number complete service was demonstrated since the pilot could, via the URD, select all system options. He could (1) get a preflight briefing by saying the word "briefing," (2) be connected to a flight service specialist by saying the word "specialist," or (3) be connected to the fast-file recording system by saying "file," "amend," or "close."

5. The use of an URD to allow the selection of system services was successful. The URD, an untrained speech recognition device that operates over switched telecommunications lines, met or exceeded its specification of understanding 95 percent of the callers when using subgroups of its vocabulary.

6. The system could not support 20 telephone lines. Eight lines were implemented for purposes of demonstrating the system. Timing measurements performed on these 8 lines indicated that the system could support 16 telephone lines.

7. Since each briefing was composed of 16 segments, message entry and update were easy and convenient for the flight service specialist.

8. Message update by the flight service specialist did not result in an interruption of service to callers using the system.

9. Automatic message composition using a small test vocabulary of 250 words was successfully demonstrated.

10. System startup and recoveries from power failures are accomplished via one-word entries at the system command console. This feature allows for nonspecialized personnel to operate and support the system unless a severe hardware malfunction occurs.

RECOMMENDATIONS

FIELD TEST.

The Mass Weather Dissemination System Exploratory Engineering Model should be prepared for a field test and then sent to a field site for test and evaluation. This test would demonstrate that the Engineering Model is not merely a "laboratory curiosity" but rather a viable candidate for Flight Service Station automation to be included in the production phase of the program.

1. The Engineering Model would require the following changes to accommodate such a test:

a. If more than 16 telephone lines are required to be serviced, then the latest version of the 7/32 operating system must be purchased. The new operating system has reportedly reduced the system overhead of its SVC1 input/output operations by 50 percent. This reduced overhead would allow the system to service more telephone lines. If 16 or less telephone lines are required to be serviced, then no changes are required.

b. Some of the fixed head-per-track discs, which are currently used for general purpose operations, must be used for voice data storage only.

c. The connection method to the specialist answering sets which are selected when the pilot says "specialist" will have to be changed. Instead of connecting to modified telephone sets, connection will have to be made via an automatic call director. The audio portion of the connection will remain unchanged; however, the control signals will need to be modified.

d. Flight plans would be recorded on existing flight plan fast-file recorders. There are currently two fast-file recorders in the system. If more than two recorders are required, new units must be purchased and modified.

2. The field test would provide answers to the following questions:

a. Is the flight service specialist's workload reduced significantly by the advent of the system?

b. Can the pilot make a go/no-go decision from the information presented by the system?

c. Does the system provide the pilot with enough information to plan a flight?

d. Is the caller's line hold-time reduced when using this system?

e. What is the general aviation public's feelings on the system overall?

f. Is the pilot provided with timely service in all areas of system utilization?

g. What is the utilization of the two new options available to the pilot? These options are specialist and fast-file.

h. Does the system provide an improved message product to the pilot?

i. Are the message products, which are presented to the pilot, representative of the most current information available?

j. Can the system operate reliably on a 24-hour-a-day continuous operation environment?

k. If the system fails, which system modules are failing and can they be made to operate reliably?

AUTOMATIC MESSAGE COMPOSITION.

The use of automatic message composition to convert text messages directly into speech at the word level seems to be a very promising method of achieving automation. With such a system in operation, most spoken messages would be generated automatically, without the use of a human operator to manually enter the messages.

The main advantages to concatenation at the word level is a vast savings in storage over a technique which involves phrase level concatenation, where words have to be stored multiple times but in different phrases. Another advantage gained by concatenation at the word level is that much less restraints are placed on the originator in formatting the weather message. A message can be generated in free text with the only constraint being that the words used in preparing the message be valid entries on the vocabulary disc.

The demonstration of automatic message composition using a small test vocabulary of 250 words indicated that this was a viable approach.

1. Continuing work in this area should include:

a. Selection of a geographical area to be serviced for purposes of providing a demonstration.

b. Selection of the geographical names required to be spoken in this area.

c. A review of the approximately 1,100 aviation/weather words required for message preparation.

d. Selection of a speaker to speak these words onto analog tape which will be used later for vocabulary building.

e. Improving the software tools required to build a vocabulary disc.

f. Demonstration of the new vocabulary fully integrated into the Mass Weather Dissemination System.

FLIGHT PLAN FILING.

It may be possible to use the Utterance Recognition Device (URD) to directly file flight plans. If this method of collecting flight plan data from the pilot was viable, the computer could easily format the data for transmission over service B. This could reduce flight service specialist workload if he was only required to check the data before sending it via service B, since he would not have to enter all of the data. If the system incorporated the appropriate route verification routines, a flight service specialist would not be required to check the flight plan data, and his role in filing flight plans from this system would be eliminated.

The filing of a flight plan utilizing the URD should be handled in a radically different way than is presently done. Only the flight plan data pertinent to the National Airspace System (NAS) should be obtained via the URD for real-time verification by the pilot. The remaining flight plan data should be recorded on the fast-file recorder since it is utilized only in case of an emergency situation. In both cases, the pilot should be prompted so that he does not forget to enter pertinent information.

This approach is required because of the tedious nature of data entry between man and machine. It is felt that the time necessary to enter all of the flight plan data via the URD would be undesirable to the user and cost-ineffective to the system.

1. Work completed in this area includes:

a. Determination of a set of words necessary for filing flight plans via the URD.

b. Collection of these spoken words and reference pattern generation for the URD.

c. Demonstration of flight plan filing utilizing the URD to collect all of the data in the flight plan.

d. Demonstration of flight plan filing utilizing the URD to collect pertinent NAS system data with the remaining data recorded on fast-file.

2. Additional work in this area should include:

a. Try to streamline the entry of flight plans by eliminating unnecessary fields.

b. Test the new set of vocabulary words to verify their recognition rate.

c. Implementation of the existing set of command words so that flight plan data may be edited by the pilot as he is making his entries.

d. Test those words in the new vocabulary that existed in the old vocabulary to see if their recognition rate has been improved by incorporation of the national data base.

e. Attain acceptance of collection of flight plan data utilizing the URD to collect pertinent NAS system data and audio-recording the remainder of the flight plan. This new approach is required because automation techniques of data collection are radically different from the existing manual system. It is felt that this approach may be accepted by the pilot population.

CODING TECHNIQUES.

Advanced techniques of coding the speech signal should be explored. The existing system encodes speech at a rate of 30,000 bits per second. Lowering of this rate will lessen the requirements for bulk storage, one of the most expensive subsystems in the system. If the encoding rate were lowered enough, it would be feasible to use commercially available semiconductor bulk memory systems instead of fixed head-per-track discs. Elimination of the rotating discs would enhance system reliability.

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APPENDIX A

VOICE ENCODER

PREFACE.

The voice encoder converts an analog voice signal into digital form. Once started by the receipt of a start command from the 7/32 processor, digital voice samples are continuously generated at a rate determined by the sample clock (6,000 hertz) until a stop command is received from the 7/32 processor. A first-in/first-out (FIFO) memory is included so that no voice samples are lost when the 7/32 processor is temporarily too busy to respond to interrupts for service from the voice encoder. The voice encoder is fabricated on an Interdata M48-013 Universal Logic Interface Module (ULIM). The logic diagram for the voice encoder is also included in this appendix.

ANALOG SIGNAL PROCESSING.

The audio input to the voice encoder is first passed through a variable gain amplifier. This amplifier serves to isolate the voice encoder from the audio signal source and provides the amplification required to increase the voice signal to levels required by the analog-to-digital (A/D) converter. This amplifier consists of an LM-741 general purpose operational amplifier operated in the inverting mode, with zero offset control.

After amplification, the analog audio signal is passed through an anti-aliasing filter. An anti-aliasing filter is a low-pass filter which prevents unwanted interference when frequency components of the analog audio signal are multiples of the sample clock (6,000 hertz).

The low-pass filter is composed of two identical sections. Each section consists of two UAF-41 Universal Active Filter networks which are stagger-tuned to achieve a sharp rolloff. The overall characteristics of the filter are a gain of one, a Q-factor of one, a break frequency of 2,800 hertz and a rolloff of 48 decibels per octave. The output of the anti-aliasing filter is connected to the sample-and-hold amplifier.

An SHM-1C-1 integrated circuit is the sample-and-hold amplifier. At the start of a conversion cycle, the sample clock GSC21 goes high and fires a one-shot which generates the signal HOLD1. The output of the sample-and-hold amplifier does not change when HOLD1 is high. HOLD1 remains high for 11 microseconds, which is slightly longer than the time required for a data conversion cycle. When HOLD1 returns to its low state, the output of the sample-and-hold amplifier follows the input once again.

ANALOG-TO-DIGITAL CONVERTER.

The A/D converter used by the voice encoder consists of a DAC-76 companding digital-to-analog (D/A) converter (conforms with Bell System u-255 companding law), an LM-311 comparator, a 2502 successive approximation register, an REF-01 reference source, one D-type flip-flop and two two-input exclusive OR gates. Specific connections of the parts which make up the companding analog to digital converter

are located on the A/D converter page of the voice encoder logic diagram which is included in this appendix. This document also contains a detailed timing diagram of the companding analog to digital converter, as well as a timing diagram of the voice encoder in general.

An A/D conversion cycle is initiated by the sample clock GSC1. The start pulse STARTO for the analog-to-digital converter must be synchronized with the A/D clock (ADCLK1), a 1.125 megahertz clock which is generated by back-to-back connected one-shots. This synchronization is accomplished by the QUE flip-flop, which is set by the coincidence of the sample clock GSC1 and the A/D clock ADCLK1. After the analog-to-digital converter has been started, the end of convert signal EOC1 goes low, clearing the QUE flip-flop. Ten microseconds later, EOC1 goes high, indicating that a conversion cycle is complete.

The first step of the encoding process is the determination of the sign bit. At the start of the conversion cycle, the flip-flop is set to its low state. Since this low is connected to the encode/decode pin of the DAC-76, no current flows into the encode outputs, and the comparator is effectively disconnected from the DAC-76. Thus, the comparator indicates the polarity of the input signal. After the sign bit has been determined, it is latched into the successive approximation register's (2502) most significant bit position and the flip-flop is set to its high state so that the DAC-76 is connected to the comparator for the remainder of the conversion cycle.

The A/D converter can now determine the magnitude of the input signal. The signal is just compared to S1111111, where S is the sign bit. Bits are converted with a successive removal technique until a decision has been made for each bit. Successive removal of bits is required for this analog-to-digital conversion scheme because current is drawn from the summing mode rather than being sourced to it.

For positive signals, the output of the comparator is the correct answer for each successive decision. Since current flows in the opposite direction for negative signals, the comparators output must be inverted. The exclusive-OR gate tied to the comparators output performs the required inversion when a negative signal is being digitized. Thus, the proper coding of all ones for full-scale inputs and all zeroes for zero value inputs is maintained for both positive and negative signal inputs. Positive 5 volts and negative 5 volts are the positive and negative full-scale inputs, respectively.

VOICE SAMPLE CLOCKS.

The timing of the voice encoder is controlled by the voice sample clock which is generated on the voice encoder module. The basic clock is a 6.57 megahertz crystal controlled Pierce oscillator. The 6.57 megahertz clock is squared by a Schmitt trigger and is then divided by a 12-bit presettable counter to achieve a nonsymmetrical square wave at 12,000 hertz. This technique was employed so that the sample clock could be easily changed. During the early days of this project, experimentation was performed with the sample clock rate.

The 12-bit counter is initially set to a predefined constant. This constant is chosen to yield a 12,000 hertz clock at the output of the counter chain. The most significant bit must initially be set to a 1. As the counter counts, the most significant bit eventually gets set from a 1 to a 0. The most significant bit

of the counter, changing state from a 1 to a 0, loads the counter once again with the constant, and the process repeats itself. The 12,000 hertz signal is referred to as the 2X clock and is divided by a flip-flop to achieve a symmetrical 6,000 hertz clock.

The 6,000 hertz clock is fed to two chains of three one-shots. One one-shot chain fires on the leading edge of the 6,000 hertz clock and the other fires on the trailing edge of the 6,000 hertz clock. Each one-shot chain generates a sample clock and its associated sample strobe. The purpose of the two timing chains is to distribute the processor workload among the 22 voice boards so that all 22 boards do not interrupt the processor simultaneously.

The first one-shot in the chain generates a 3,600 nanoseconds (ns) pulse which becomes the sample clock. The second generates a delay of 450 ns which triggers the third one-shot. The third one-shot generates a pulse of 450 ns which becomes the sample strobe. Thus, the sample clock is 3,600 ns long, and the sample strobe is 450 ns long and begins 450 ns after the sample clock. The sample clocks are buffered to other voice boards in the system via 7437 NAND buffers with appropriate RC networks at the outputs.

Most of the voice boards are remote from the encoder module. A cable distributes the voice sample clocks and strobes to these boards. The cable connector at the remote end of the cab contains Schmitt triggers to reconstitute the signals and buffers (7437) to distribute the signals within the remote chassis.

VOICE SAMPLE MUTIPLEXING.

Any time that the voice encoder is enabled, voice samples are constantly being generated at a rate determined by the sample clock. Three 5-bit voice samples are multiplexed into one 16-bit word, which is usually referred to as a halfword in Interdata literature. The 16-bit word is temporarily stored in a first-in/first-out (FIFO) memory before being sent to the 7/32 processor. The multiplexer is a set of 74175 and 7474 latches which are controlled by the sample counter.

The sample counter is initially set to 00 by the start command. The first sample clock after the start command starts the A/D converter with its leading edge and increments the sample counter to a count of 01 with its trailing edge. Approximately 6 microseconds after the trailing edge of the sample clock, the A/D conversion is complete and EOC1 goes high. This generates CLK01 which loads the 5-bit voice sample just generated into the most significant portion of the 16-bit data word (bits 1-5).

The next sample clock starts the A/D converter and increments the sample counter to 10. At the end of the A/D conversion, EOC1 goes high and generates CLK11. CLK11 loads the 5-bit voice sample into the middle position (bits 6-10) of the 16-bit data word.

The third sample clock starts the A/D converter and increments the sample counter to 11. The count 11 is detected by the sample counter logic, which clears the sample counter to a count of 00. When the A/D conversion is complete, EOC1 goes high and generates CLK21. CLK21 loads the 5-bit voice sample into the least significant (bits 11-15) of the 16-bit data word.

The 16-bit data word now contains three 5-bit voice samples and is complete. CLK21 is converted to become SHIFTIN1, which enters the 16-bit data word into the FIFO memory.

The multiplexer is now set up to multiplex the next three voice samples. This process continues until the voice encoder is disabled via a stop command.

VOICE ENCODER FIFO MEMORY.

The voice encoder is enabled for operation after it receives a start command (COT040) from the 7/32 processor. The start bit clears the data overflow flip-flop, clears the sample counter to a count of 00, resets the FIFO memory, and interrupts the processor for service.

When the FIFO memory is reset, the data output ready signal (DOR1) goes low, indicating that there are no more data in the memory and the data input ready signal (DIR1) goes high, indicating that there is room in the memory for new data.

A 16-bit data word is completed when CLK21 clocks the third 5-bit voice sample into the 16-bit data word latch. CLK21 also generates the signal SHIFTIN1 which strobes the newly formed 16-bit data word into the input end of the FIFO memory.

Since there are no other data in the FIFO memory, the FIFO memory control logic moves (bubbles) the 16-bit data from the input end of the FIFO memory to the output end of the FIFO memory, one location at a time. The last location of the FIFO memory is called the output position, and this is where the 16-bit data word eventually resides.

When data are located in the output position, the signal data output ready (DOR1) goes high, indicating that data are available for the 7/32 processor. DOR1 generates the signal (SATNO) which interrupts the 7/32 processor for service.

Since the outputs of the FIFO memory are connected to the Universal Interface Logic Module data input lines DIN000 through DIN150 via buffer amplifiers, the 16-bit data word at the output of the FIFO memory is acquired by the 7/32 processor when the 7/32 processor executes a read instruction to the voice encoder. The execution of the read instruction causes the data request signal (DRG1) to be generated. DRG1 becomes SHIFTOUT1. The trailing edge of SHIFTOUT1 causes the next 16-bit word in the FIFO memory to be brought to the output location (if one exists).

If there are more data in the FIFO memory, the data output ready signal (DOR1) goes low, momentarily indicating that the data at the output position are not yet ready. If additional data were available in the FIFO memory, the signal DOR1 would immediately go high again, thus interrupting the 7/32 processor for another service request. If there were no additional data in the FIFO memory, DOR1 would stay low indicating to the processor that no more data were available. When data become available, DOR1 goes high and interrupts the 7/32 processor service.

The process of multiplexing 5-bit voice samples into 16-bit data words and the placing of the 16-bit data words into the FIFO memory continues until the voice encoder receives a stop command from the 7/32 processor. The stop command stops the multiplexing of voice samples into 16-bit data words by removing the sample clock (GSC1). Interrupts are still generated until the FIFO memory is empty.

STATUS AND COMMAND DATA.

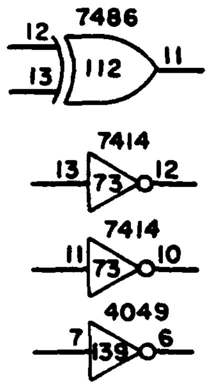
During the normal course of operation, the voice encoder and 7/32 processor must communicate control information to each other. The processor transfers control data to the voice encoder via the command byte while the voice encoder transfers control data to the processor via the status byte.

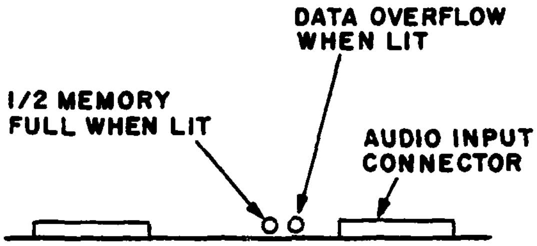
Three of the eight bits of the command byte are used by the voice encoder. The two most significant command bits, bits 0 and 1 (Interdata usually refers to them as bits 8 and 9 due to their position on the multiplexer bus) are used to arm and enable interrupts. These bits are fully discussed in the ULIM manual. Bit 4 (COT040) is the start bit. This bit clears the voice encoder logic and starts the process of encoding voice data.

Three status bits are used. Bit 3 (SIN030) is connected to the start bit (COT040) so that the programmer can verify that the voice encoder has been started. Bit 5 (SIN050) is the data overflow indicator. This bit is set when the FIFO memory is full (DIRO is true) by the clock CLK11. This tells the programmer that voice data have been lost.

The final status bit is the busy bit (SIN040). BUSYO (SIN040) is generated by inverting DORO. When the FIFO memory has no data available for the 7/32 processor, the busy bit is low to indicate that fact. The busy bit controls the data transfer between the voice encoder and the 7/32 processor.

SPARE LOGIC





FRONT VIEW OF ENCODER BOARD

132 741	121 POTS	110 POTS	99 DAC76	88 7474	77 74107	66 74175
133 UAF41	122 SPARE	111 2502	100 RESISTOR REF 01	89 7402	78 7408	67 7474
134 UAF41	123 RESISTOR FOR 133&134	112 7486	101 7474	90 7404	79 7408	68 74175
135 UAF41	124 RESISTOR FOR 135&136	113 SHM IC1	102 LM311	91 7420	80 7408	69 74175
136 UAF41	125 R/C FOR 114	114 74123	103 74123	92 R/C 81&103	81 74123	70 7474
137 SPARE	126 SPARE	115 7437	104 74123	93 R/C 82&104	82 74123	71 7474
138 4012	127 40105	116 40105	105 40105	94 40105	83 74197	72 74197
139 4049	128 40105	117 40105	106 40105	95 40105	84 74197	73 7414
140	129	118	107	96 75492	85 75492	74 75492
141	130	119	108	97	86	75
142	131	120	109 RESISTOR FOR CLOCK	98 R/C FOR CLOCK	87	76

81-56-A-1

FIGURE A-1. STUFFING CHART FOR ENCODER

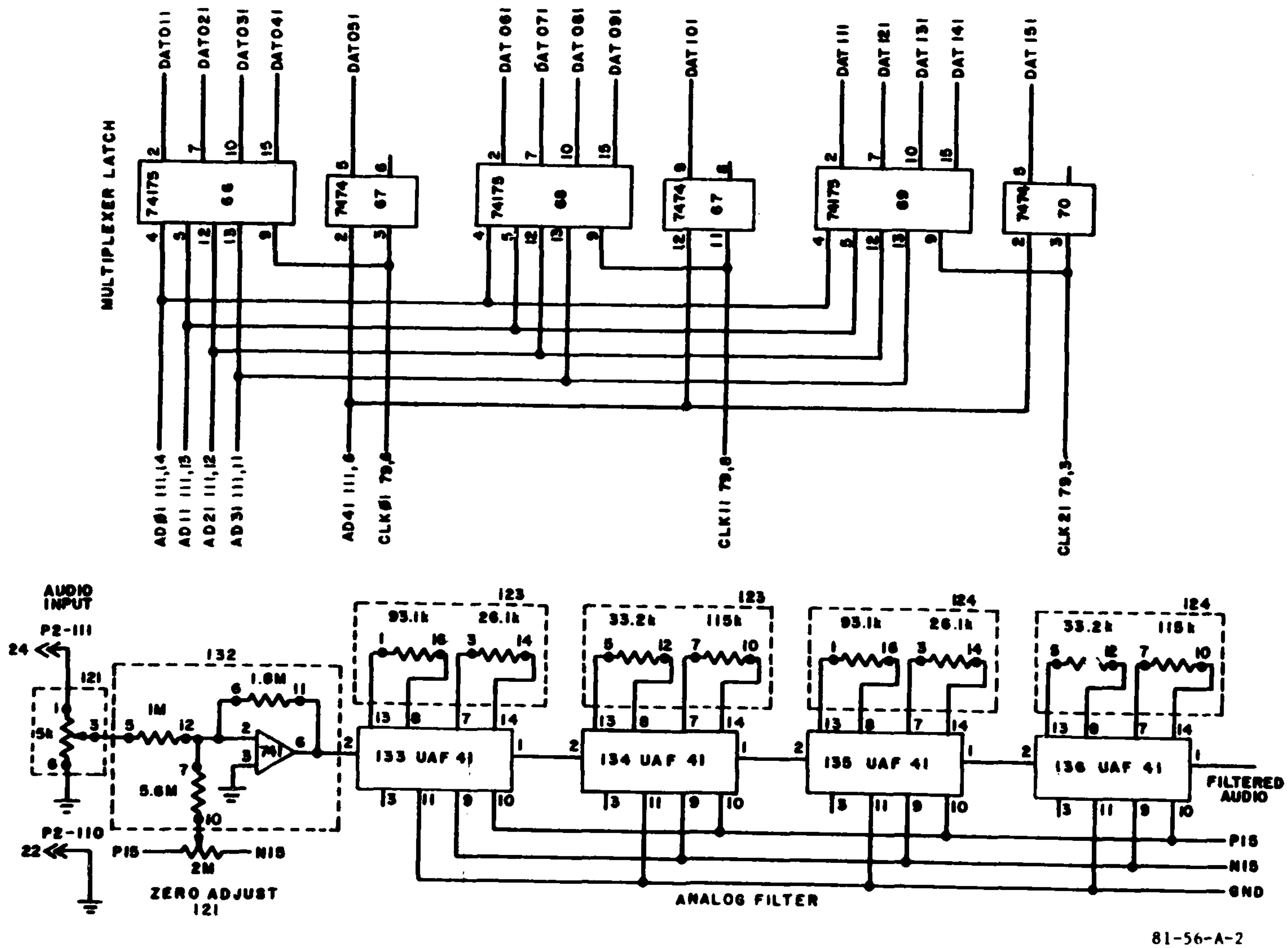
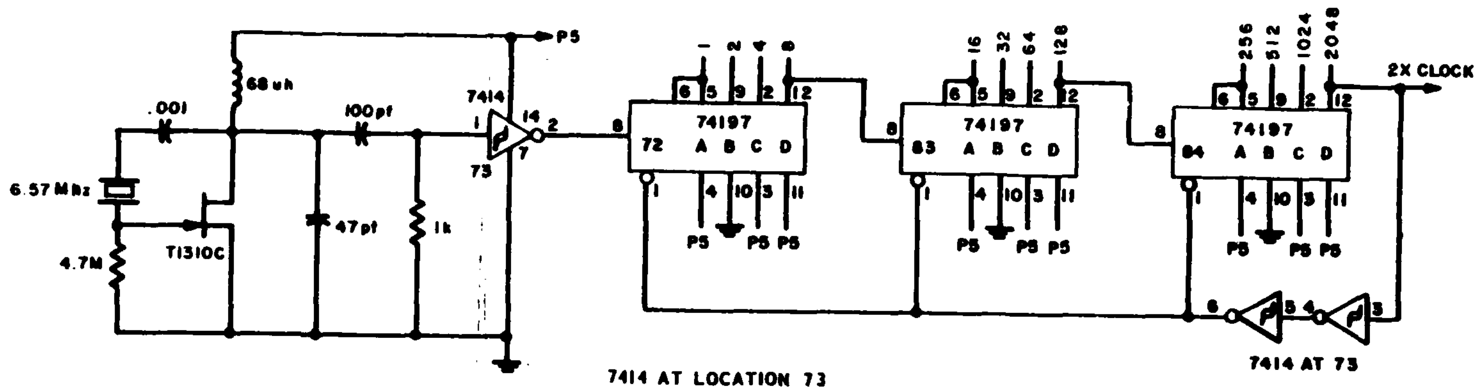


FIGURE A-2. MULTIPLEXOR LATCH AND ANALOG FILTER



COUNTERS 74197 OR 8291
SCHMITT 7414

CRYSTAL OSCILLATOR 6.577770 Mhz
AT ROOM TEMP.

2X CLOCK IS 12,003 Hz

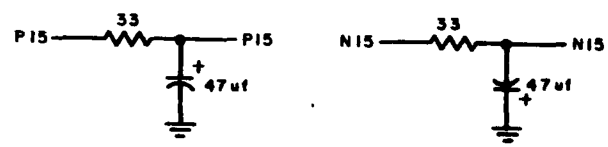
MINIMUM FREQUENCY 3211.8 Hz

FREQUENCY CALCULATION:
DESIRED FREQUENCY 12,000 Hz
6.577 Mhz ÷ 12000Hz = 548.15
548 = 512 + 32 + 4
PUT GROUNDS (LOGIC 0) ON
THESE PRESET INPUTS.
FINE TUNING MAY BE REQUIRED
APPROPRIATELY.

THE MSB (2048) MUST ALWAYS
BE PRESET TO A ONE

7414 AT LOCATION 73
DUE TO A WIRING MISTAKE, THE
47pf CAPACITOR IS PARALLEL WITH
THE 1k RESISTOR.

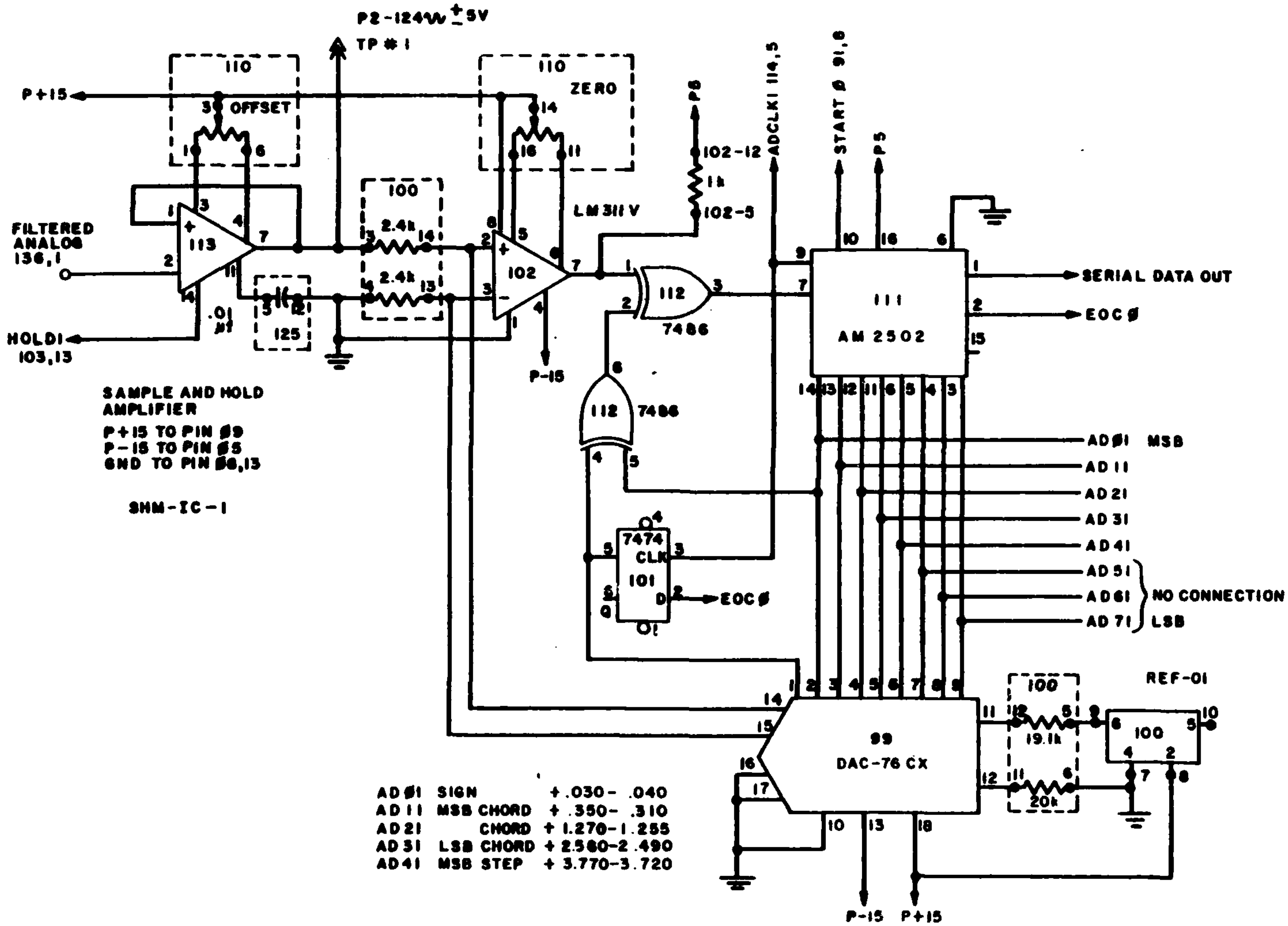
PARTS LOCATED AT POSITION 64
FOR FET, CRYSTAL AND OTHER
DISCRETE PARTS



PLUS AND MINUS FIFTEEN VOLT FILTERS
PARTS LOCATED AT 53 AND 61

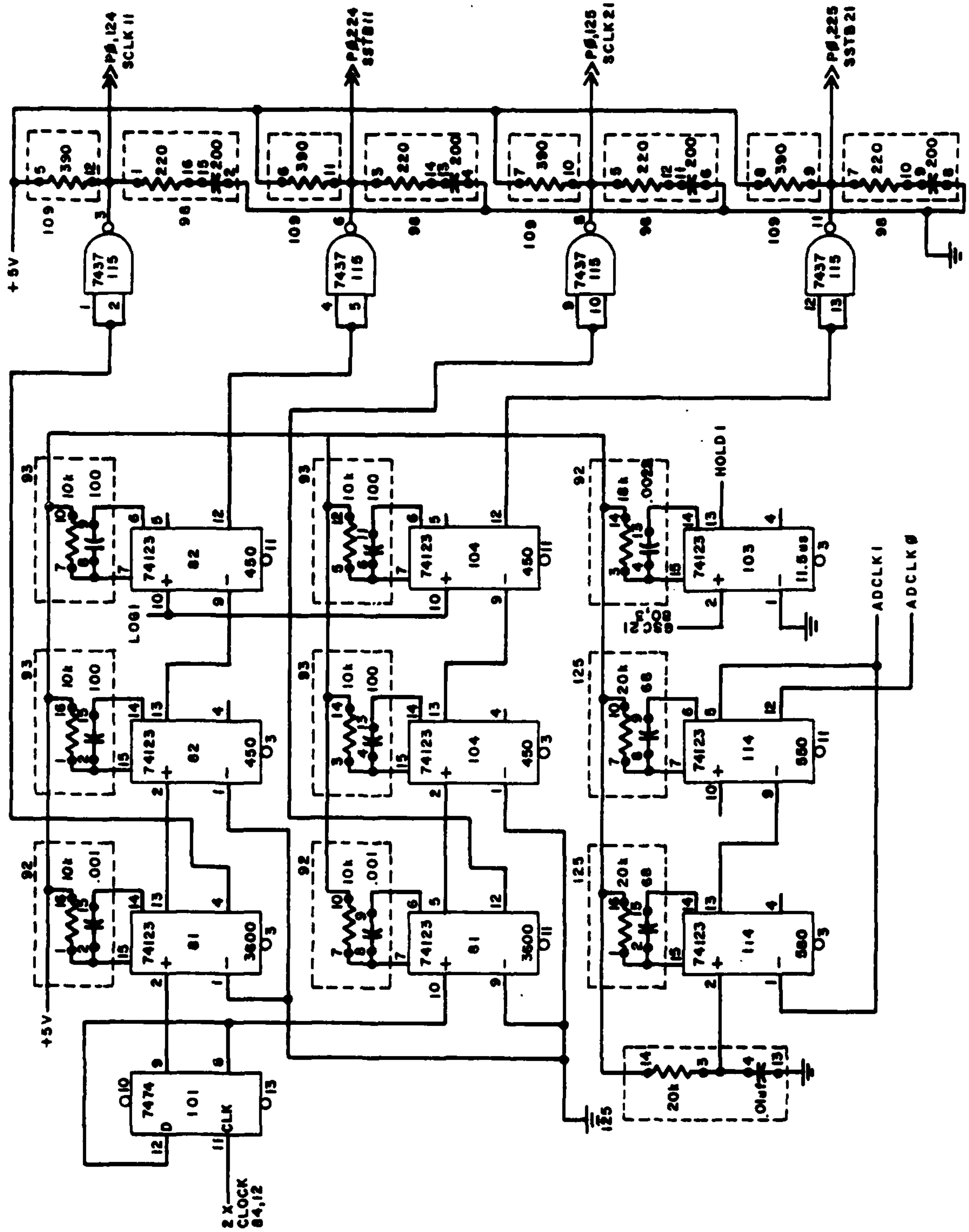
81-56-A-3

FIGURE A-3. CRYSTAL CONTROLLED CLOCK, DIGITALLY VARIABLE



81-56-A-4

FIGURE A-4. ANALOG-TO-DIGITAL CONVERTER



81-56-A-5

FIGURE A-5. SAMPLE CLOCKS AND ANALOG-TO-DIGITAL CLOCK

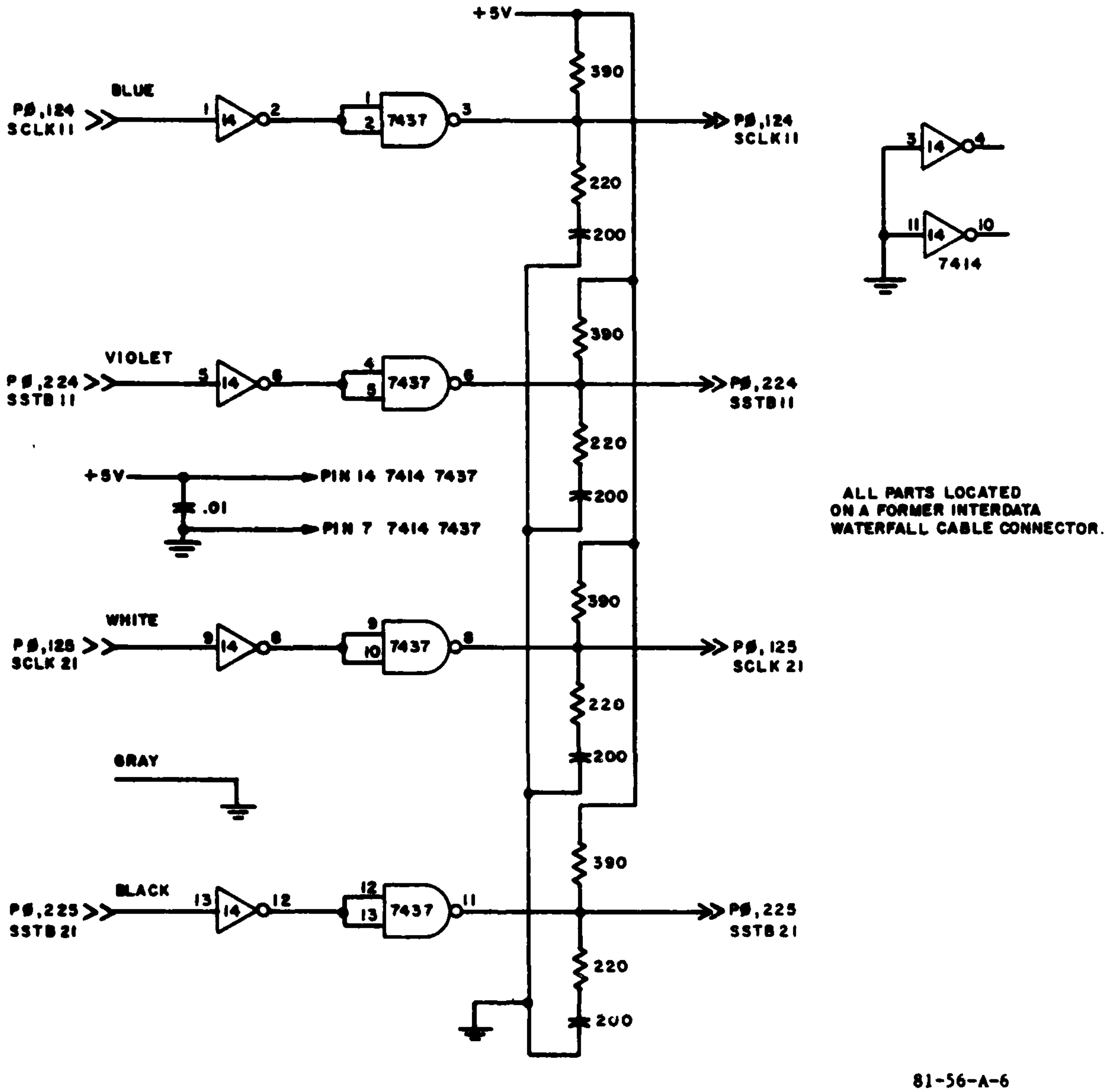


FIGURE A-6. SAMPLE CLOCK DISTRIBUTION

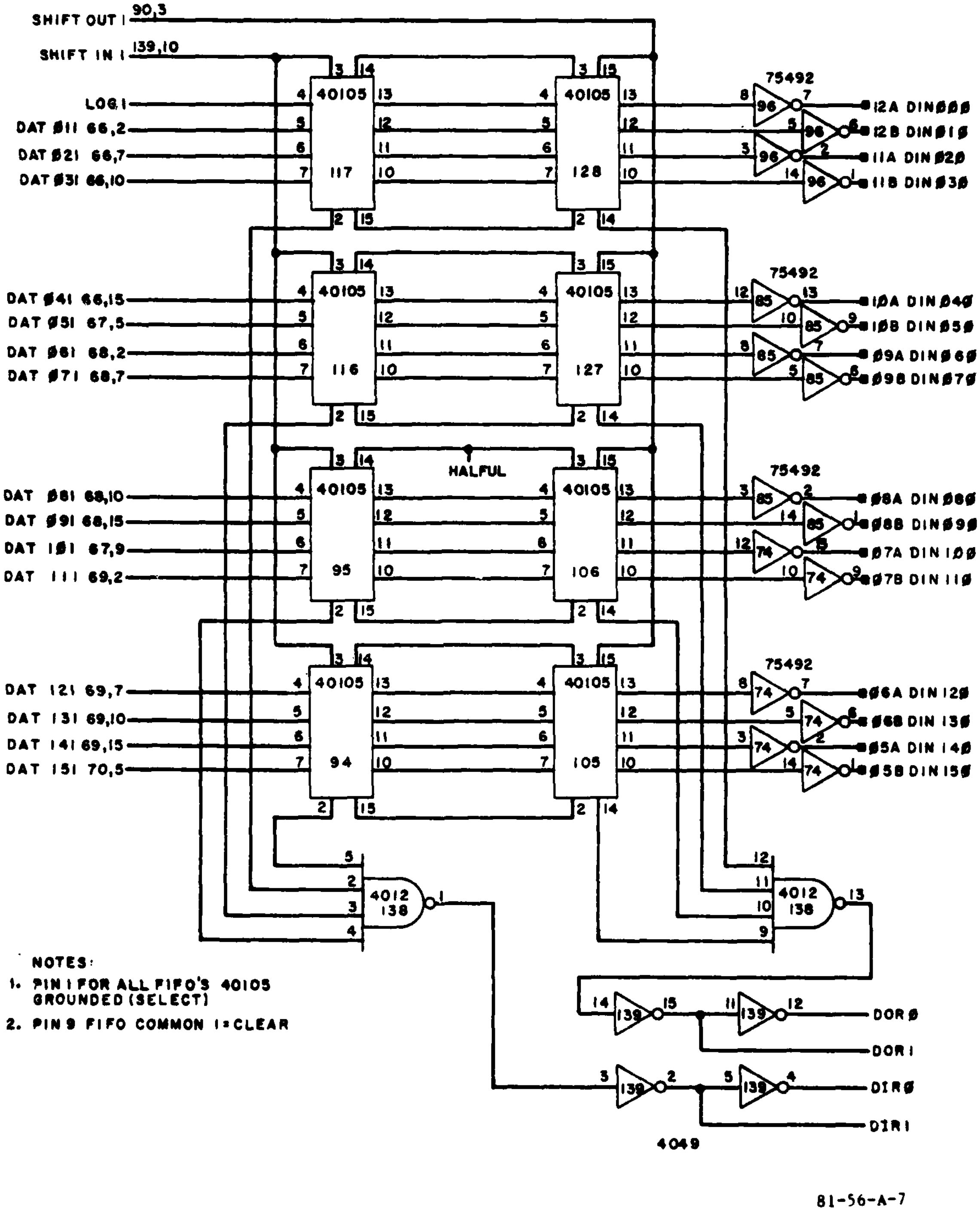
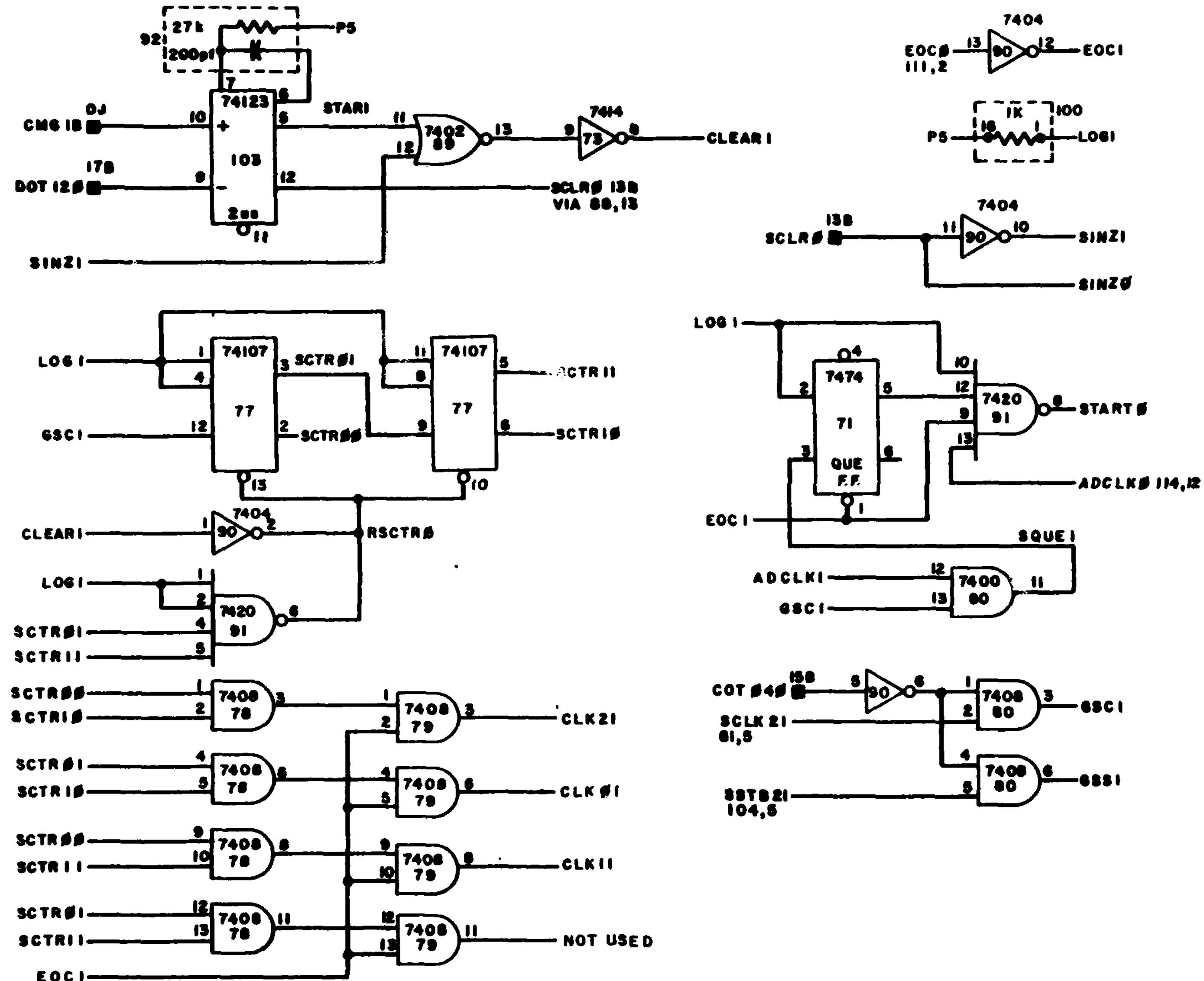
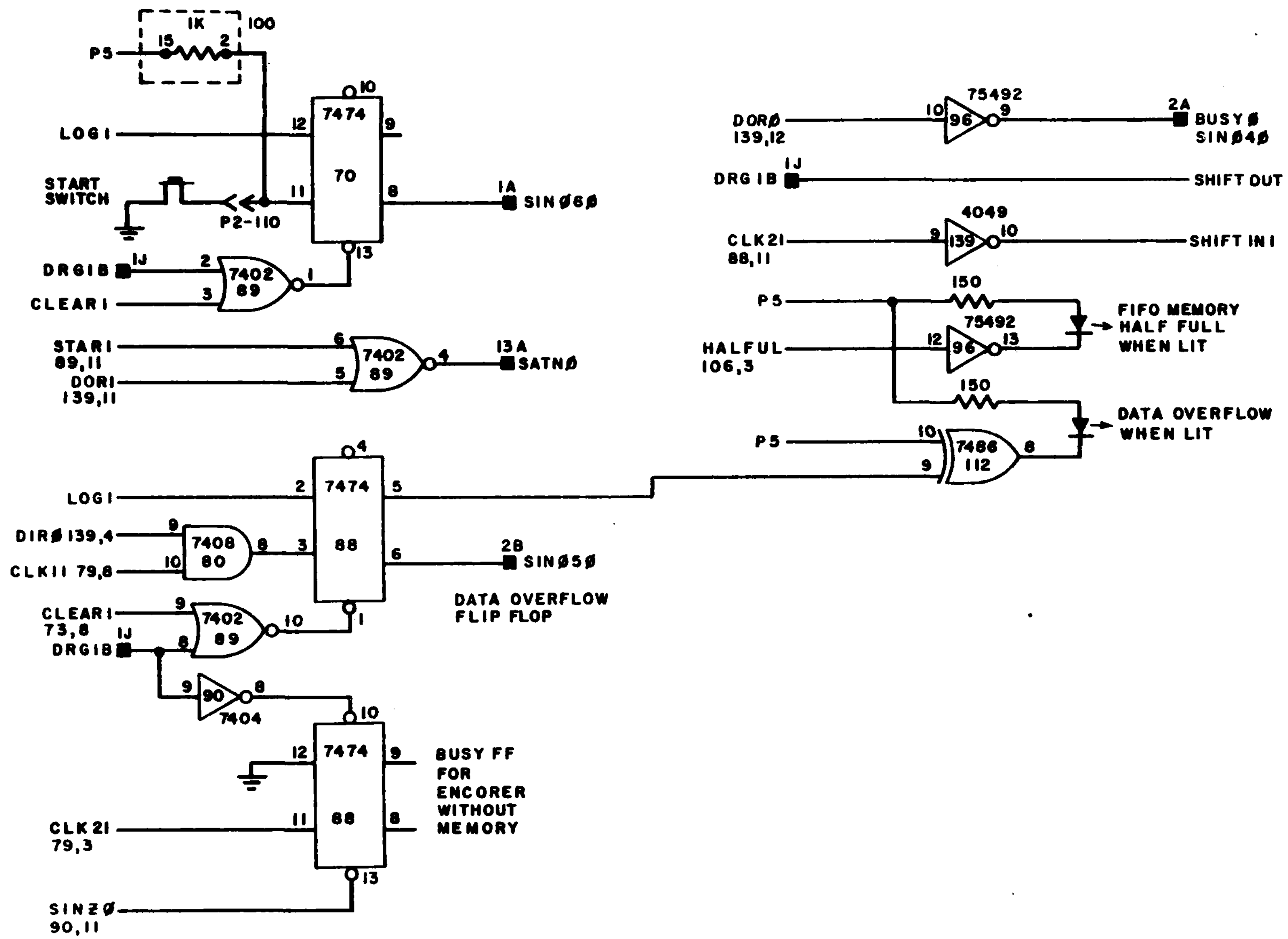


FIGURE A-7. ENCODER FIRST-IN/FIRST-OUT MEMORY



81-56-A-8

FIGURE A-8. ENCODER CONTROL LOGIC (SHEET 1 OF 2)



81-56-A-8

FIGURE A-8. ENCODER CONTROL LOGIC (SHEET 2 OF 2)

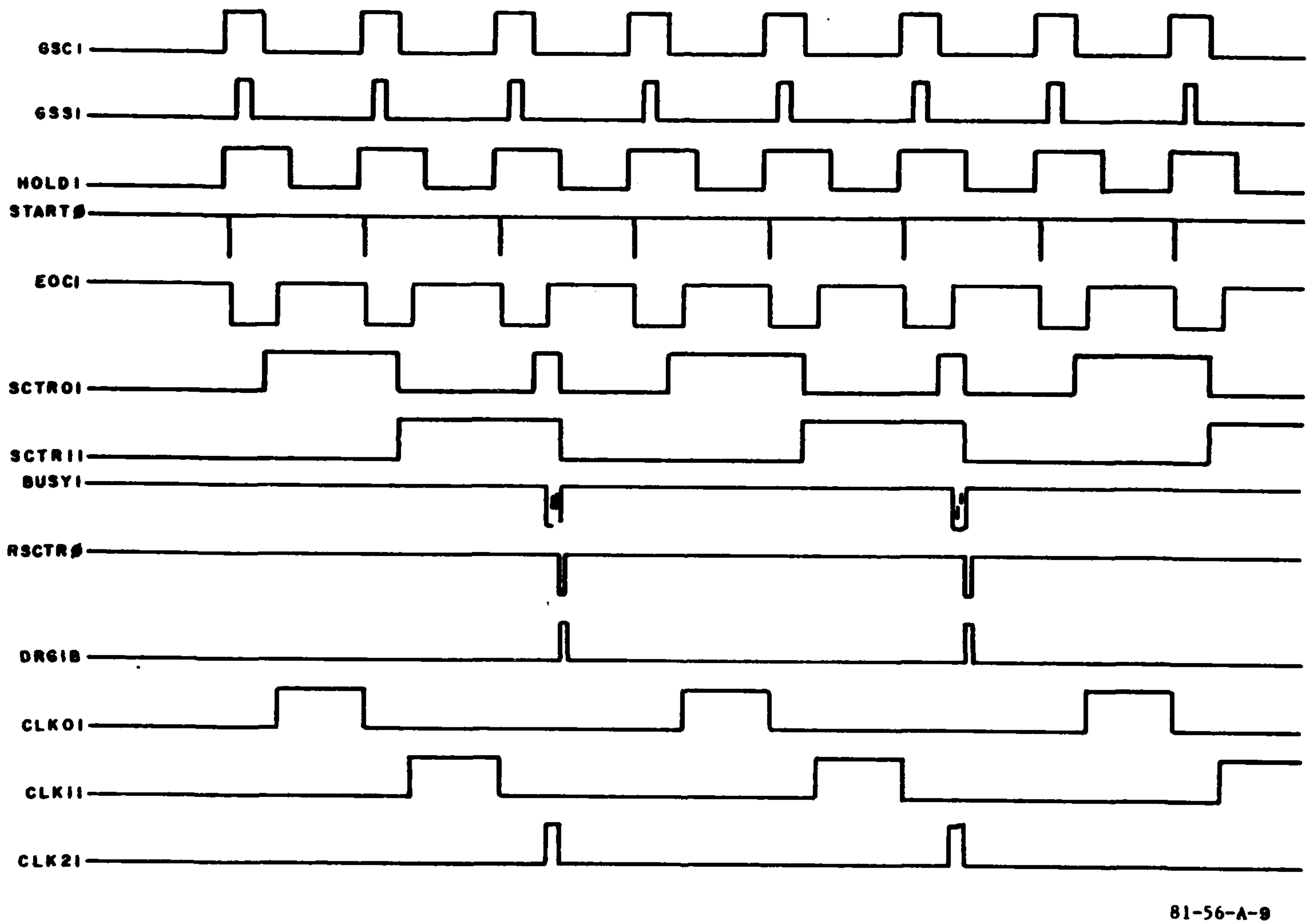


FIGURE A-9. ENCODER TIMING NO FIRST-IN/FIRST-OUT MEMORY

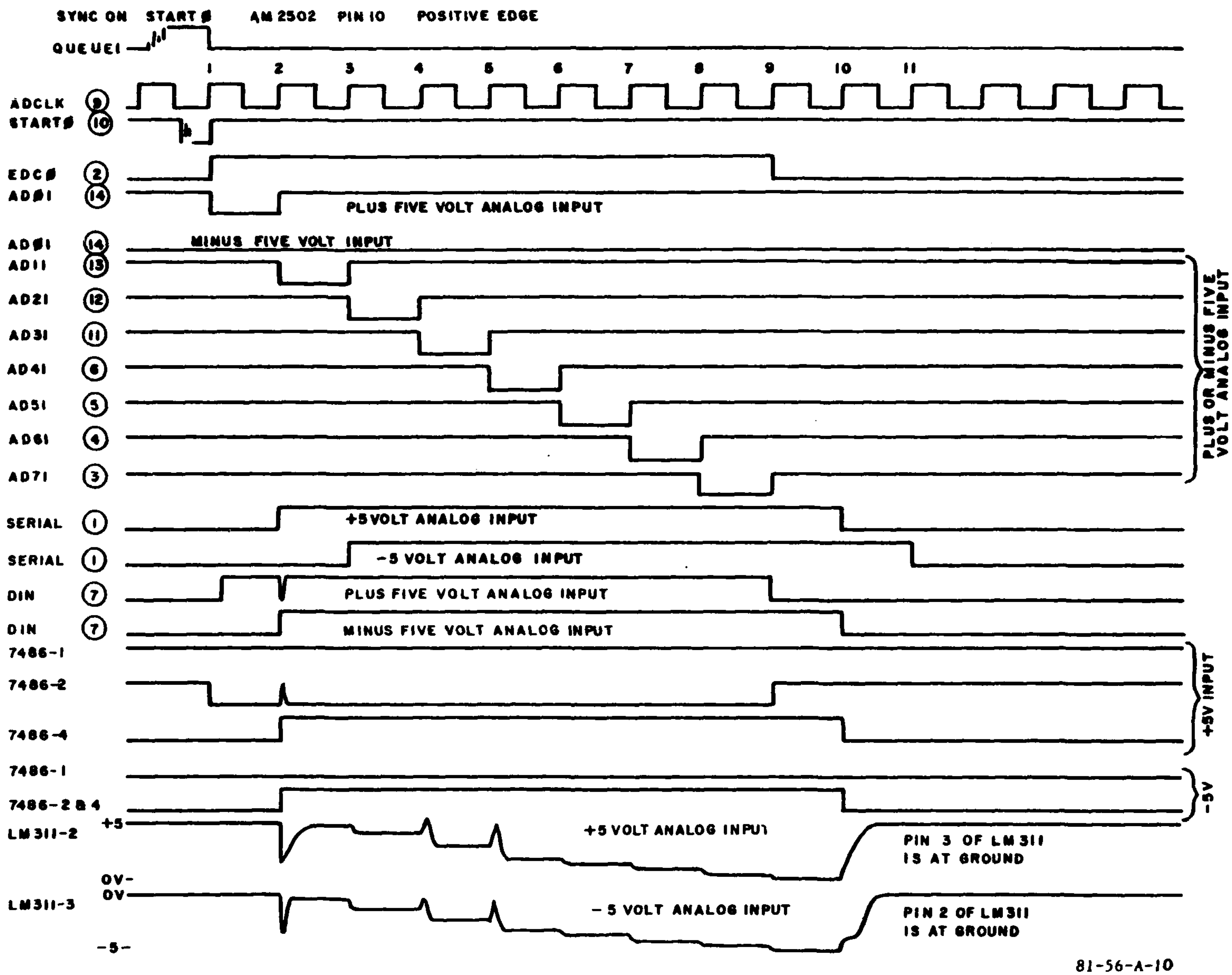


FIGURE A-10. ANALOG-TO-DIGITAL TIMING

APPENDIX B

VOICE DECODER

PREFACE.

The voice decoder converts digitally encoded voice data received from the 7/32 processor into a speech signal. The speech signal is amplified and played out via a loudspeaker which is mounted on the manual voice entry console. The voice decoder is fabricated on an Interdata number M48-013 Universal Logic Interface Module (ULIM). The logic diagram for the voice decoder is also included in this appendix.

STARTUP LOGIC.

The voice decoder begins operation after receipt of a start command from the 7/32 processor. The start command fires the upper one-shot on the control logic page which generates the signal SINZ1. SINZ1 clears the sample counter to 00, clears the data overflow flip-flop, resets the first-in/first-out (FIFO) memory and interrupts the 7/32 processor for service. The voice decoder is now set up to generate speech when it receives data from the 7/32 processor.

Data from the 7/32 processor are presented to the voice decoder via the processor interface part of the ULIM. When data are available, the signal DAG1 goes high. At this point, the 16-bit voice data word would be placed in the FIFO memory.

FIFO MEMORY.

The FIFO memory is fabricated with eight CD40105 FIFO memory integrated circuits which are series and parallel connected to form a FIFO memory of 32 16-bit words. The operation of this memory is exactly as it is described for one of the individual FIFO integrated circuits.

Since the FIFO memory had just been reset by the start command, it is ready to accept 16-bit voice data words. The reset pulse causes DIR1 to go high, indicating that the memory is ready to accept data, and DOR1 to go low, indicating that there are no voice data in the memory.

When the 7/32 processor sends a 16-bit voice data word to the voice decoder, the signal DAG1 goes high which in turn causes the signal SHIFTIN1 to go high. SHIFTIN1 causes the voice data word to be placed in the FIFO memory. During this time, the signal DIR1 goes low to indicate that the FIFO memory is busy and cannot accept more data. After SHIFTIN1 goes low once again and if there are still vacancies in the FIFO memory, the signal DIR1 goes high. This signal fires the lower one-shot on the control logic page, which interrupts the processor for service by generating the interrupt signal SATNO.

This process of interrupting the processor for more voice data words and the storage of voice data words in the FIFO memory continues until the FIFO memory is full. When the FIFO memory is full, the signal DIR1 remains low after SHIFTIN1 becomes low. DIR1 remains low until there is a vacancy in the FIFO memory, then it goes high and interrupts the 7/32 processor for another voice data word to fill the vacancy.

After a voice data word has been placed in the FIFO memory, the FIFO memory's self-contained control logic moves the data to the output location of the FIFO memory. When voice data appear at the output location of the FIFO memory, the signal DOR1 goes high. Speech generation begins at this point because DOR1 enables the sample clock GSC1 and sample strobe GSS1.

DEMULTIPLEXER.

Since the FIFO memory output lines are connected to the demultiplexer input lines, the three 5-bit voice samples of the 16-bit voice data word at the output location of the FIFO memory are successively separated and stored in the voice sample latch. The demultiplexer is controlled by the sample counter which was reset to 00 by the start command. The demultiplexer begins to operate when the sample clocks are enabled.

The leading edge of the first sample clock (GSC1) does not affect the sample counter. Since the sample counter is in the 00 state, the demultiplexer is set to take the first 5-bit voice sample (bits 1-5 of the 16-bit data word) and store them in the sample latch. The sample strobe (GSS1) occurs 1/2 microsecond later and stores the 5-bit voice sample in the voice sample latch. The trailing edge of the sample clock occurs 2.5 microseconds after the sample strobe and increments the sample counter to 01.

The count 01 of the sample counter causes the demultiplexer to select the middle 5-bit voice sample (bits 6-10) from the 16-bit voice data word. The second sample strobe stores this voice sample in the voice sample latch. The trailing edge of the second sample clock increments the sample counter to 10.

During the third sample clock period, the last voice sample (bits 11-15) of the 16-bit voice data are selected by the demultiplexer. The third strobe signal (GSS1) stores the voice sample in the voice sample latch and also generates the signal SHIFTOUT1, which requests another 16-bit voice data word from the FIFO memory. The trailing edge of the third sample clock (GSC1) increments the sample counter to 11.

The sample counter state of 11 is detected, and a reset signal (RSCTR0) is generated which clears the sample counter to its initial state of 00. The demultiplexer is now set up to demultiplex the next 16-bit voice data word into its component three 5-bit voice samples. The demultiplexing process continues until the FIFO memory becomes empty (DOR1 is low).

DIGITAL TO ANALOG CONVERTER.

The outputs of the voice sample latch are connected to the inputs of the digital-to-analog (D/A) converter, which produces a voltage that is proportional to the digital value of the sample latch. An LM 741 operational amplifier, a DAC-76 companding D/A converter, and an REF-01 reference source are integrated circuits which make up the D/A converter.

The REF-01 and its 19,100-ohm resistor form a reference source for the DAC-76 which determines the DAC-76 step size. The DAC-76 is a companding (conforms with Bell System u-255 companding law) D/A converter which produces a current output which is proportional to its digital inputs. Although the DAC-76 is an 8-bit converter, only the 5 most significant bits are required in this application. An LM-741 general purpose operational amplifier is configured as a current to voltage converter which takes the current output of the DAC-76 D/A converter and converts it to the voltage range which is required.

The voltage at the output of the LM-741 is proportional to the digital value of the sample latch. Since the sample latch is updated with digital voice samples at a rate determined by the sample clock (6,000 hertz), a voice waveform is generated.

AUDIO OUTPUT.

The voice waveform, at this point, contains sharp discontinuities which are caused by the digitizing process. These sharp discontinuities are removed from the speech signal by a low-pass filter.

The low-pass filter is composed of two identical sections consisting of two UAF-41 Universal Active Filter networks which are stagger-tuned to achieve a sharp rolloff. The overall characteristics of the filter are a gain of one, a Q-factor of one, a break frequency of 2,800 hertz and a rolloff of 48 decibels per octave.

The speech signal at the output of the low-pass filter drives an LM-380 2-watt audio amplifier integrated circuit. The LM-380 amplifies the speech signal so that it can drive a loud speaker which is mounted on the manual voice entry console.

STATUS AND COMMAND DATA.

Three status bits convey data concerning the operational status of the voice decoder to the 7/32 processor. Computer programs can make decisions based on this status information.

The first status bit (SIN030) informs the processor that the voice decoder has been enabled for operation. SIN030 is obtained by a connection to the start command bit (COT040). When the voice decoder has been enabled, COT040 is true and, thus, SIN030 is true.

The second status bit (SIN040) is called the busy bit. This bit controls the rate at which the processor sends data to the voice decoder. DIR1, the data input ready signal from the FIFO memory, controls the busy bit. When DIR1 is high, the FIFO memory can accept voice data words, and DIR1 being high causes BUSY0 to be high, which indicates to the processor that the voice decoder can accept voice data. When DIR1 is low, the FIFO memory cannot accept data, and BUSY0 is low in order to inform the processor that the voice decoder is busy and cannot accept more data at this time.

The third and final status bit (SIN050) is the data overflow bit. The first sample clock (GSC1) after the FIFO memory becomes empty (DOR0 is high) sets the data overflow flip-flop. This status informs the program that speech generation by the voice decoder has been interrupted.

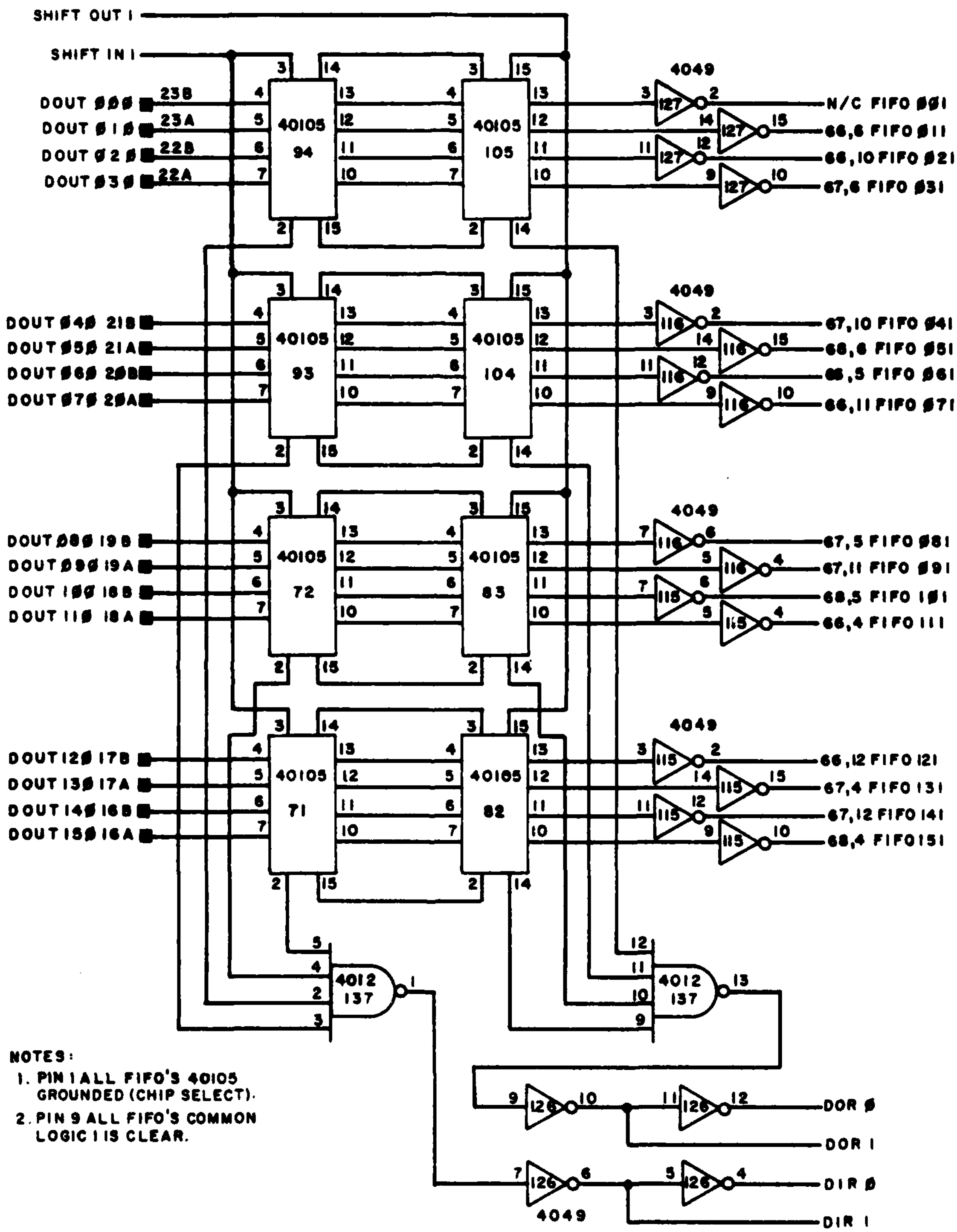
Three command bits allow the processor to control the voice decoder. The two most significant command bits of the command byte, bits 0 and 1 (Interdata usually refers to these bits as bits 8 and 9 due to their position on the multiplexer bus) are used to arm and enable voice decoder interrupts for service. The function and operation of these command bits are fully described in the Interdata M48-013 ULIM manual.

Bit 4 (12), COT040, is the start bit. The start bit clears the voice decoder control logic so that the voice decoder may begin generating a speech signal as soon as voice data are received from the processor.

132 LM 380	121 PARTS POT	110 OP-02	99 DAC-76	88 7408	77 74175	66 74153
133 PARTS DAC & LM 380	122 PARTS FOR UAF 111	111 UAF-41	100 REF 01	89 7404	78 7474	67 74153
134 PARTS FOR 74123	123 74123	112 7474	101 7402	90 7410	79 74107	68 74153
135 UAF 41	124 UAF RESISTORS	113 UAF 41	102 UAF 41	91 UAF RESISTORS	80	69
136 R/C PARTS	125 SKT	114 SKT	103	92	81	70
137 4012	126 4049	115 4049	104 40105	93 40105	82 40105	71 40105
138 SKT	127 4049	116 4049	105 40105	94 40105	83 40105	72 40105
139	128	117	106	95	84	73

81-56-B-1

FIGURE B-1. STUFFING CHART FOR DECODER



NOTES:
1. PIN 1 ALL FIFO'S 40105 GROUNDED (CHIP SELECT).
2. PIN 9 ALL FIFO'S COMMON LOGIC 1 IS CLEAR.

81-56-B-2

FIGURE B-2. DECODER FIRST-IN/FIRST-OUT MEMORY

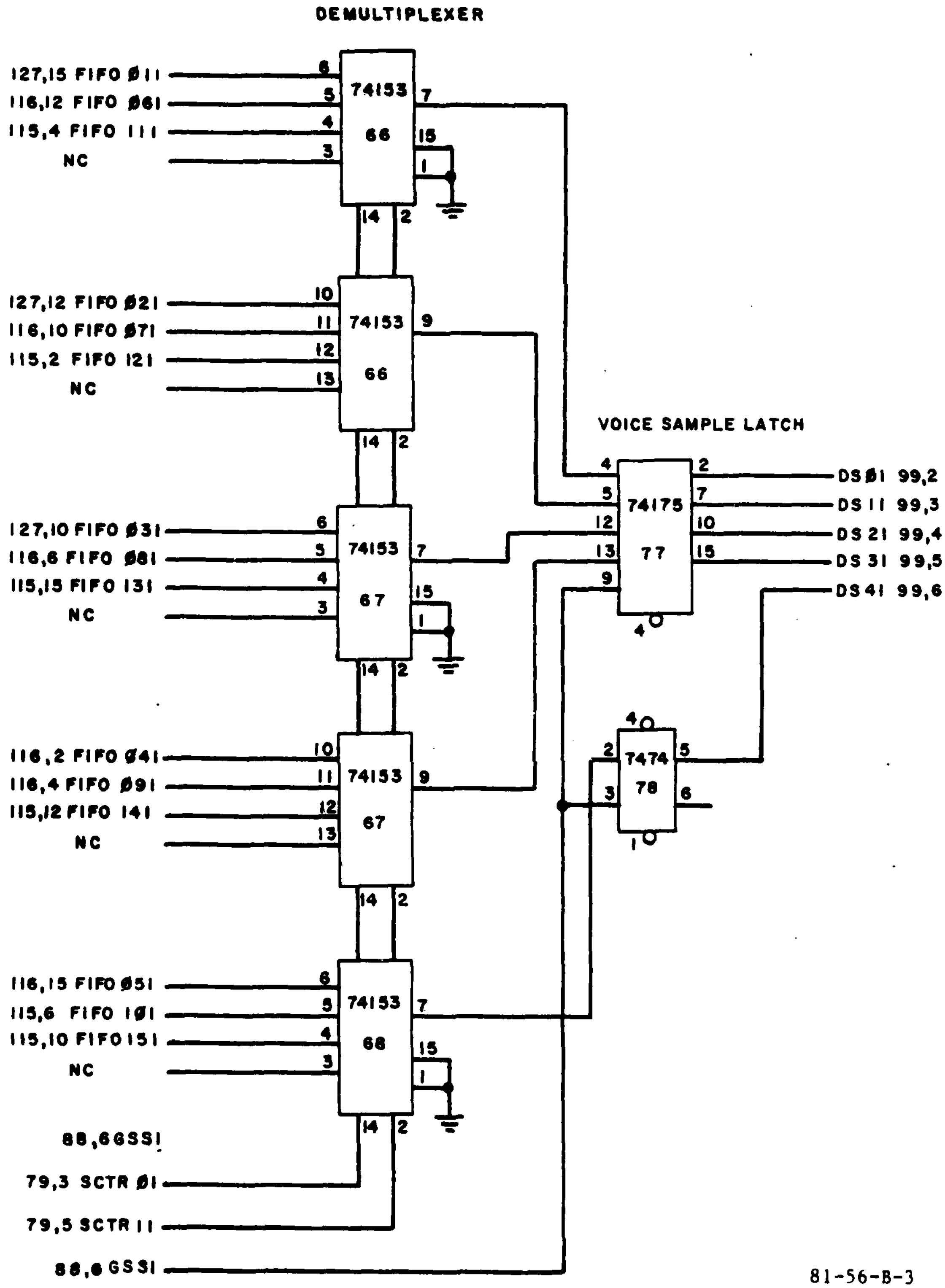


FIGURE B-3. DEMULTIPLEXER LOGIC

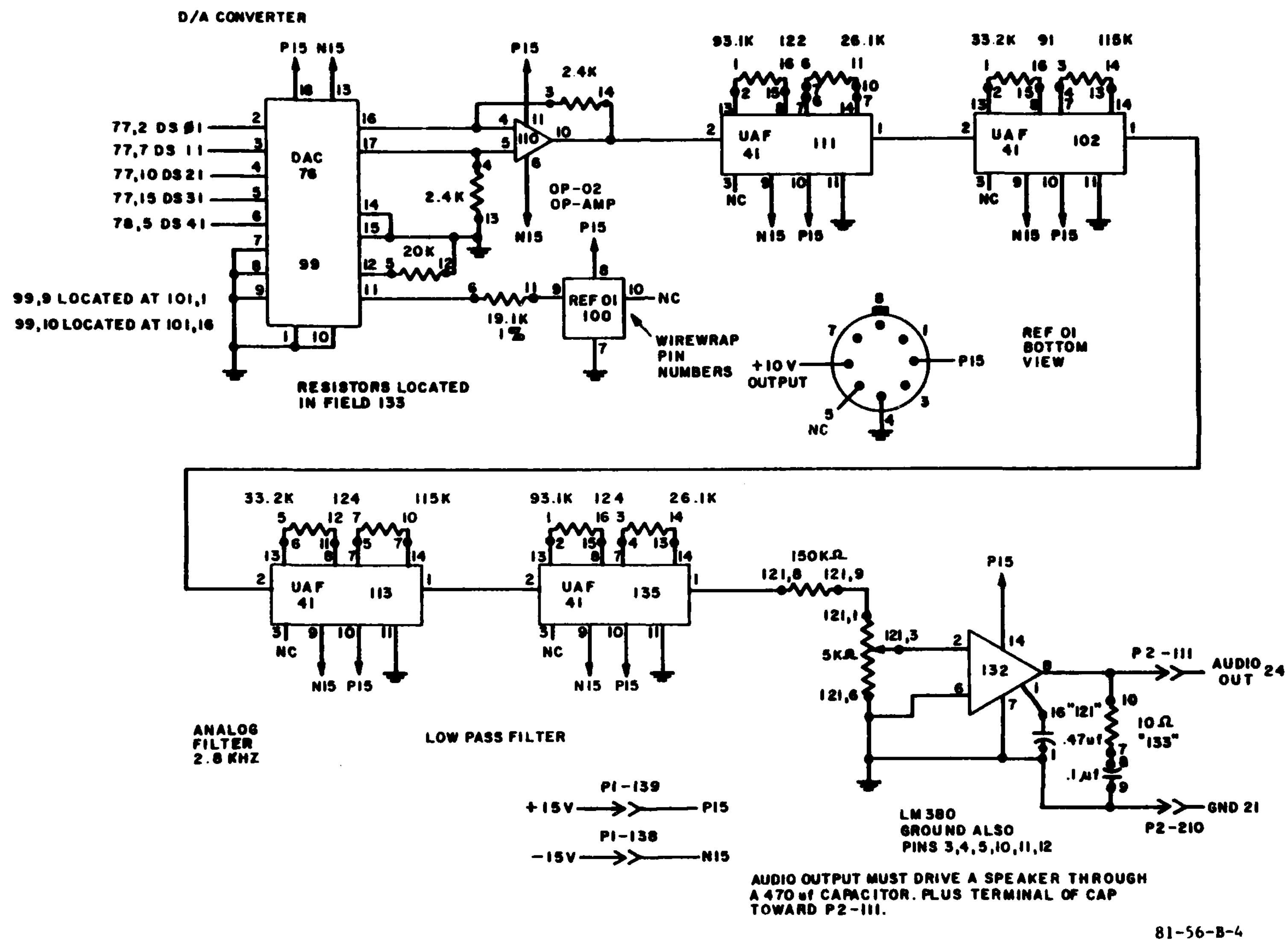


FIGURE B-4. ANALOG SECTION

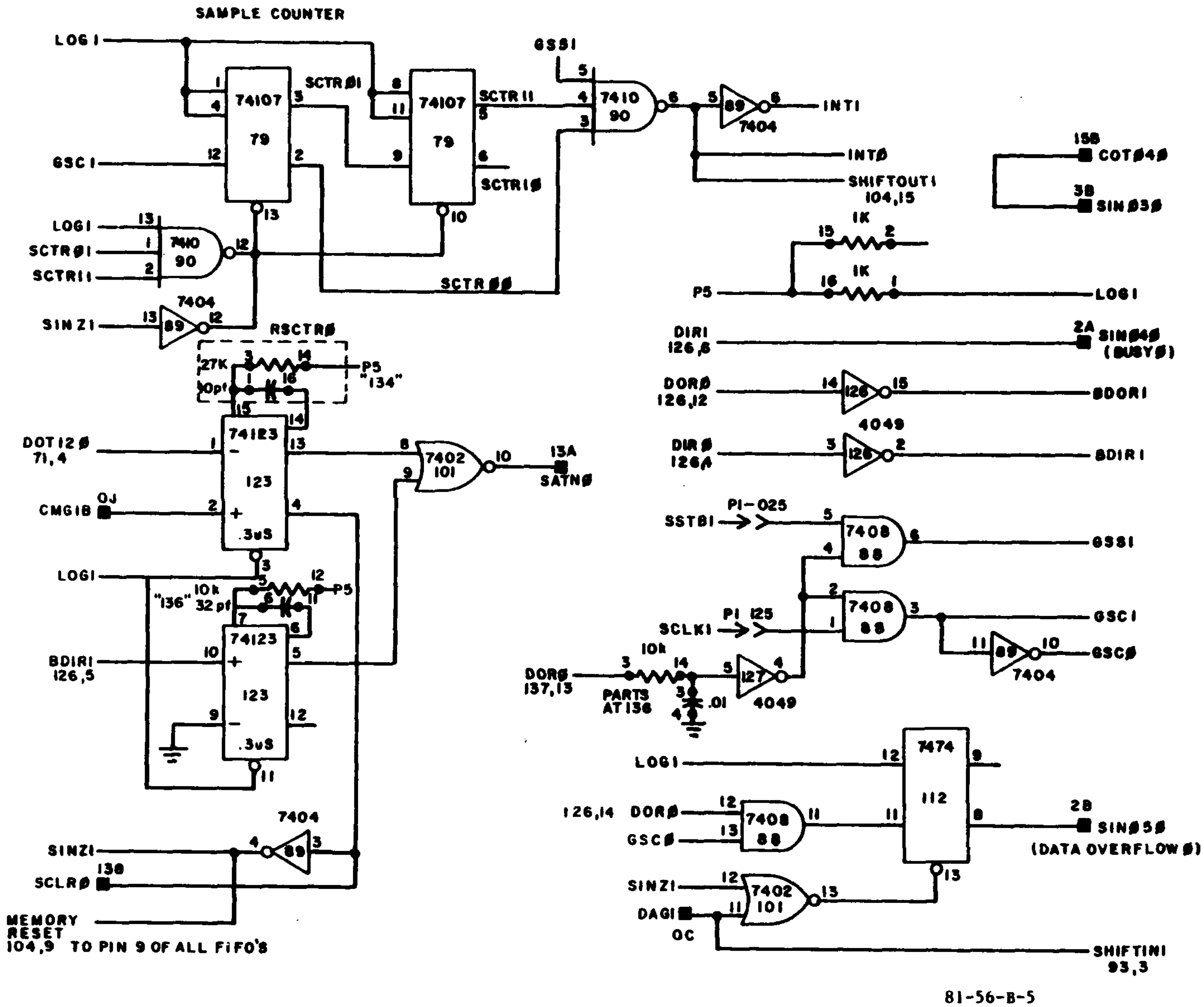
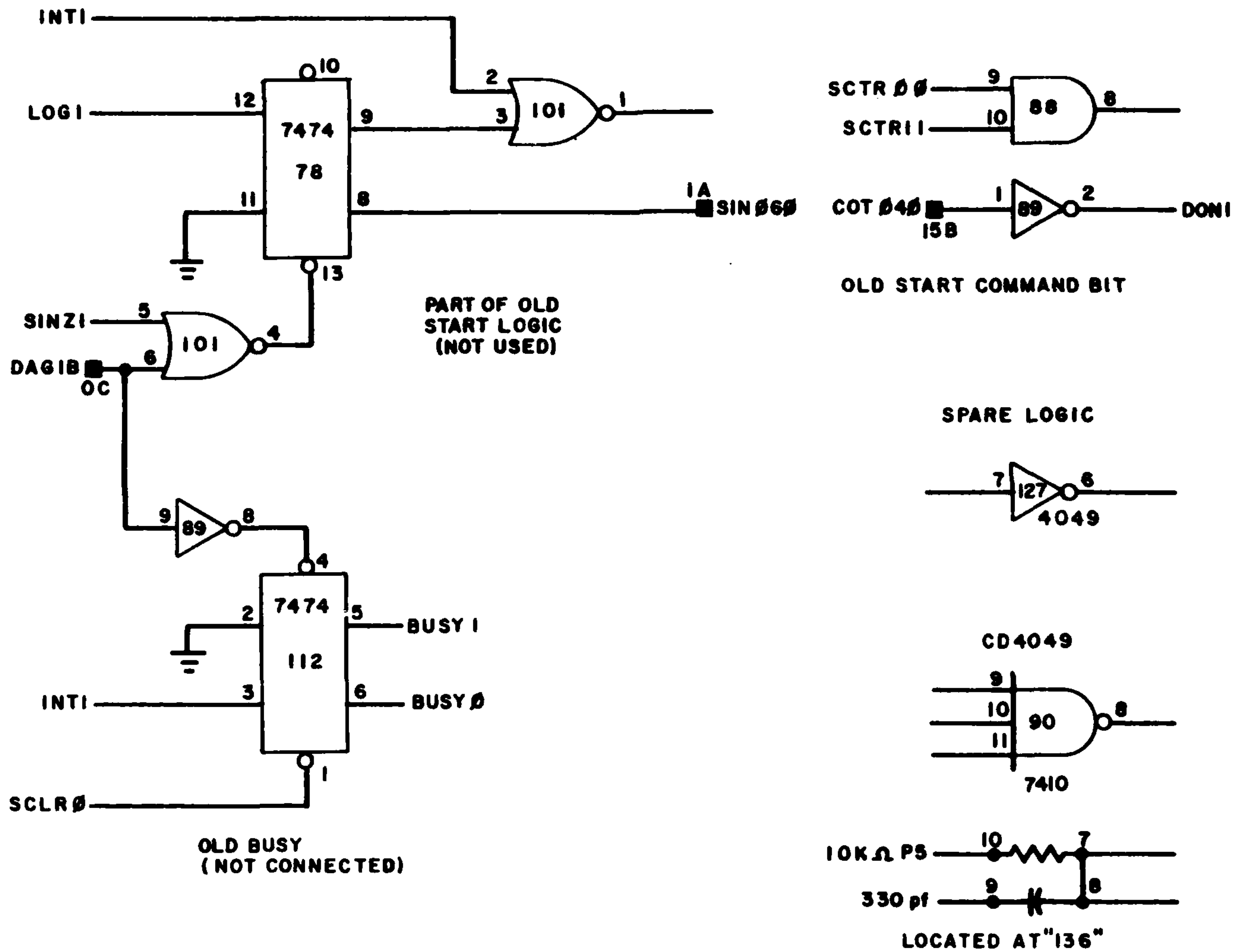


FIGURE B-5. DECODER CONTROL LOGIC



81-56-B-6

FIGURE B-6. OLD CONTROL LOGIC (NOT USED)

APPENDIX C

TELEPHONE VOICE DECODER

PREFACE.

The telephone voice decoder is designed to interface with the telephone company via a Bell System CDH voice connecting arrangement and CBF connecting arrangement. This system element contains all of the logic required for the reconstruction of digital voice samples into speech for transmission over telephone lines, the logic to handle telephone control signals, such as ring and pickup, and the logic to control the internal memory. The telephone voice decoder is fabricated on a MDB Universal Interface Logic Module (ULIM) number MDB-48-013-16-01. The logic diagram for the telephone voice decoder is also included in this appendix.

STARTUP LOGIC.

The telephone voice decoder is enabled for operation after it receives a start command (CM131) from the 7/32 processor. Actual voice decoding does not begin until the first of two memory buffers is filled with voice data. The start bit sets the data overflow flip-flop to 1, clears the memory busy flip-flop, resets the read request and read mode flip-flops, clears the write request and write mode flip-flops, and sets the steer flip-flop to 0 (STEER1 = low). The start command also sets the voice address register (VAR) to 600, the processor address register (PAR) to 000, clears the 3-ring flip-flop, clears the telephone ring counter to 0, and clears the sample counter.

INTERNAL MEMORY.

The memory of the telephone voice decoder consists of 16 21L02 1024 by 1 memory chips arranged to form a memory of 1024 16-bit data words. The memory control logic divides the memory into two buffers of 512 16-bit data words each. When STEER1 is low, the processor is set to fill the lower buffer with new voice data and the data in the upper buffer are converted into speech and played out over the telephone line. When STEER1 is high, the processor fills the upper buffer with new voice data and the data in the lower buffer are converted into speech and played out over the telephone line.

Data transfers to the telephone voice decoder are handled in a different manner from data transfers to the voice encoder and voice decoder. Data are transferred to the voice encoder and voice decoder via the multiplexer bus; whereas, data are transferred to the telephone voice decoder via a selector channel. Use of the selector channel provides for data transfers direct from memory without processor intervention, thus reducing system overhead.

MEMORY CONTROL LOGIC.

Data flow to and from the memory is determined by the memory address registers. These registers are presetable and may be incremented upon command. The processor address register (PAR) points to the next memory location that will be updated by the processor. The voice address register (VAR) points to the next memory location whose data will be converted to voice and played out over the telephone line. Memory addresses from the VAR and PAR are multiplexed to the memory address lines of the memory elements.

In the following discussion, data will be referred to as being read from the internal memory or written into the internal memory. During a read operation or read cycle, the telephone voice decoder is reading data out of the internal memory and converting it into speech. During a write operation or write cycle, the 7/32 processor is writing new digital voice data into the internal memory.

MEMORY WRITE CYCLE.

A write cycle is started when the signal DAG1 is received from the processor. This signal indicates that the processor has sent a 16-bit voice data word to the telephone voice decoder. DAG1 sets the write request flip-flop. If a read cycle is not in progress, the write flip-flop is set (WFF1 is high) and the memory timing is enabled.

Memory timing is provided by three one-shots. The first one-shot (T11) fires on the lead edge of WFF1 and has a duration of 400 ns. This allows the memory address lines time to settle. At the end of this time, the second one-shot (T21) fires. This 500-ns pulse becomes the write pulse WTO (WTO is low). This pulse also becomes CPAR1, whose trailing edge increments the processor address register (PAR). At the end of this time, the third one-shot (T31) fires for 100 ns. This pulse becomes RWFFO and clears the write request and write mode flip-flops.

MEMORY READ CYCLE.

When the demultiplexer has separated the third and final 5-bit voice sample from the 16-bit data word, it requests another 16-bit data word from the memory by generating INT1. INT1 sets the read request flip-flop which in turn sets the read mode flip-flop if a write cycle is not presently in progress.

Memory timing is once again initiated. T11 is fired by read mode (RFF1) going high. T11 allows the memory address lines to settle. T21 fires at the end of T11 and becomes CPAR1. The leading edge of CPAR1 strobes the 16-bit data word into the memory data register and the falling edge increments the VAR. At the conclusion of T21, T31 is fired and clears the read request and read mode flip-flops.

MEMORY CONTENTION LOGIC.

It is possible that the read mode and write mode flip-flops are set simultaneously. When this happens, the memory contention logic gives priority to the processor by allowing the write cycle to continue. The read mode flip-flop is cleared, but the read request flip-flop remains set. At the conclusion of the write cycle, a read cycle is initiated.

If a read request occurs during a write cycle, the read request flip-flop is set and no further action is taken. When the write cycle is completed, the read mode flip-flop is set and a read cycle is in progress. A write request during a read cycle works similarly.

MEMORY OPERATION.

The state of the memory buffers is constantly monitored. When STEER1 is high and VAR091 goes high, the lower buffer is empty (BOE1 is high). The upper buffer is full (B1F1 is high) when PAR101 is high and STEER1 is high. When STEER1 is low.

the upper buffer is empty (B1E1 is high) when VAR101 is high, and the lower buffer is full (B0F1 is true) when PAR091 is high. PAR091, PAR101, VAR091, and VAR101 are the most significant bits of the processor address register and voice address register, respectively.

Suppose that the upper buffer is full of new voice data and those data are being taken from the lower buffer and being converted into speech. When the lower buffer becomes empty, STEER1, the memory control flip-flop, goes from high to low. Any change in STEER1 causes the delta steer pulse (DSTEER1) to be generated.

In this instance, DSTEER0 causes the VAR to point to the first location of the upper buffer and the PAR to point to the first location of the lower buffer. DSTEER1 interrupts the processor for service.

As speech is being generated, voice data are successively removed from the upper buffer. It takes approximately 1/4-second for the telephone voice decoder to empty a buffer. The lower buffer is filled within a few milliseconds after the selector channel begins transferring data to the telephone voice decoder.

Since the processor data transfers to the telephone voice decoder are so fast, the processor's buffer is always filled with new voice data before the voice data buffer is emptied. A few miliseconds after the steer flip-flop has changed state, the processor has filled the lower buffer (B0F1 is true); and about 250 milliseconds later, the voice buffer becomes empty (B1E1 is true).

When both B0F1 and B1E1 are true, the steer flip-flop is set (STEER1 is true). This change in STEER1 causes DSTEER1 to be generated which interrupts the processor for service, sets the PAR to point to the first location of the upper buffer, and sets the VAR to point to the first location of the lower buffer.

Speech is generated from data in the lower buffer while new voice data are being transferred to the upper buffer. A few milliseconds after DSTEER1, the upper buffer is full (B1F1 is true). The voice buffer empties approximately 250 milliseconds after the DSTEER1 signal (B0E1 is true).

When both B0E1 and B1F1 are true, the steer flip-flop is reset (STEER1 is false) The change of STEER1 causes DSTEER1 which interrupts the processor for service, sets the VAR to point to the first location of the upper buffer, and sets the PAR to point to the first location of the lower buffer.

Speech is generated from data in the upper buffer, while new voice data are being transferred to the upper buffer. This process of swapping buffers continues until the processor stops sending voice data or until the telephone voice decoder receives a stop command from the processor.

DEMULTIPLEXER.

Each 16-bit voice data word contains three 5-bit voice samples. The demultiplexer separates the voice samples from the 16-bit word and successively applies them to the digital-to-analog (D/A) converter.

The demultiplexer is controlled by the sample counter, which is initially set to a count of 00. The leading edge of the first sample clock (GSC1) does not affect the sample counter. The demultiplexers are set to take the first 5-bit voice sample

(bits 1 through 5 of the 16-bit data word) by virtue of the sample counter being in the 00 state. The sample strobe (GSS1) occurs 1/2 microsecond later and latches bits 1 through 5 of the 16-bit data word into the voice sample latch. The trailing edge of the sample clock occurs 2.5 microseconds after the sample strobe and increments the sample counter to 01.

The count 01 of the sample counter selects the middle 5-bit voice sample from the 16-bit word (bits 6 through 10). The second sample clock strobes bits 6 through 10 into the voice sample latch. The trailing edge of the sample clock increments the sample counter to 10.

During the third sample clock period, the least significant bits (bits 11 through 15) of the 16-bit data word are strobed into the voice sample latch. The strobe signal (GSS1) becomes INT1, which requests another 16-bit data word from memory at this time. The trailing edge of the sample clock increments the sample counter to 11.

The sample counter state of 11 is detected and a signal is generated (CLEAR1) which clears the sample counter to its initial count of 00. The demultiplexer is now set to demultiplex the next 16-bit voice data word into three 5-bit voice samples. This process continues until the memory becomes empty or a stop command is issued by the processor.

DIGITAL TO ANALOG CONVERTER.

The outputs of the sample latch are connected to the inputs of a D/A converter, which produces a voltage that is proportional to the digital value of the sample latch. An LM741, a DAC-76, and an REF-01 are integrated circuits which comprise the D/A converter.

The REF-01 and its 19,100-ohm resistor form a reference source for the DAC-76 which determines the DAC-76 step size. The DAC-76 is a companding (conforms with Bell System u-255 companding law) digital to analog converter that produces a current output which is proportional to its digital inputs. Although the DAC-76 is an 8-bit converter, only the five most significant bits are used in this application. An LM741 general purpose operational amplifier is configured as a current to voltage converter that takes the current output of the DAC-76 and converts it to the voltage range which is required.

The voltage at the output of the LM741 operational amplifier is proportional to the digital value of the sample latch. Since the sample latch is updated with voice samples at a rate determined by the sample clock (6,000 hertz), a voice waveform is generated.

OUTPUT STAGE.

The voice waveform, at this point, has sharp discontinuities which are caused by the quantizing process. These sharp discontinuities are removed by a low-pass filter.

The low-pass filter is composed of two identical sections, consisting of two UAF-41 Universal Active Filter networks which are stagger-tuned to achieve a sharp rolloff. The overall characteristics of the filter are a gain of one, a Q factor of 1, a break frequency of 2,800 hertz, and a rolloff of 48 decibels per octave.

The output of the last filter stage drives a telephone matching transformer (PCT-77, 600 ohms center-tapped). The secondary of the matching transformer is connected to the telephone system voice lines via a relay contact.

The relay contact is controlled by the memory overflow (MOVO) signal. When the memory no longer contains voice data, the relay contact opens, removing the telephone voice decoder from the telephone voice lines. This enhances the operation of other system options, such as fast-file, specialist, and the URD by removing the loading effects of the telephone voice decoder.

STATUS INFORMATION.

The status bits are the means by which the telephone voice decoder indicates its operational condition to the 7/32 processor. Six status bits are used to convey this information.

The busy bit (SI121), which is called EMBUSY1, controls the rate of data flow between the 7/32 processor's selector channel and the telephone voice decoder's memory. EMBUSY1 is derived from MBUSY1. MBUSY1 is made false (data transfers are enabled) when the STEER flip-flop has changed state, because at this time the processor buffer of the telephone voice decoder is not full. When the processor buffer becomes full, the MBUSY1 signal is made true, and data transfers are disabled.

MBUSY1 is false during the entire time that the processor is transferring data to the telephone voice decoder. EMBUSY1 follows MBUSY1 except that it becomes true during decoder memory write cycles. EMBUSY1 being true indicates that the telephone voice decoder is busy and cannot accept more data. When the write cycle is over, EMBUSY1 once again assumes the state of MBUSY1.

MOV1 being true indicates that there is a data overflow condition and becomes SI131. Memory overflow becomes true when the voice buffer empties before the processor buffer is refreshed. This signal, in addition to being a status bit causes the sample clock and sample strobe signals to be inhibited (speech generation halts) and disconnects the telephone voice decoder from the telephone system voice lines.

The start command bit (CM121) is also connected to SI111. This allows the 7/32 program to verify that a particular telephone voice decoder was actually turned on.

The pickup command bit (CM131) is connected to SI101. This allows the 7/32 program to verify that a pickup command was sent to a particular telephone voice decoder.

The connect signal is sent from the Bell System telephone interface circuitry. This signal becomes SI151 and indicates to the 7/32 program that a pilot is indeed connected to this particular telephone line. When the pilot hangs up in the middle of a briefing, the connect signal goes false and the 7/32 program enables that line to receive a call from the next pilot.

The final status signal is the ring signal (3RING1) which becomes SI141. The ring signal from the telephone interface is first passed through a one-shot which prevents noise spikes from being recognized as a ring. The output of the one-shot increments the ring counter which had been initialized to 0. The third ring causes the 3RING flip-flop to be set (3RING1 is true). In addition to being a status bit (SI141), the 3RING1 signal interrupts the 7/32 processor for service.

COMMAND INFORMATION.

The two most significant bits of the command byte, bits 0 and 1, (Interdata usually refers to these bits as 8 and 9 due to their position on the multiplexer bus) are used to arm and enable interrupts. These bits are fully described in the ULIM manual.

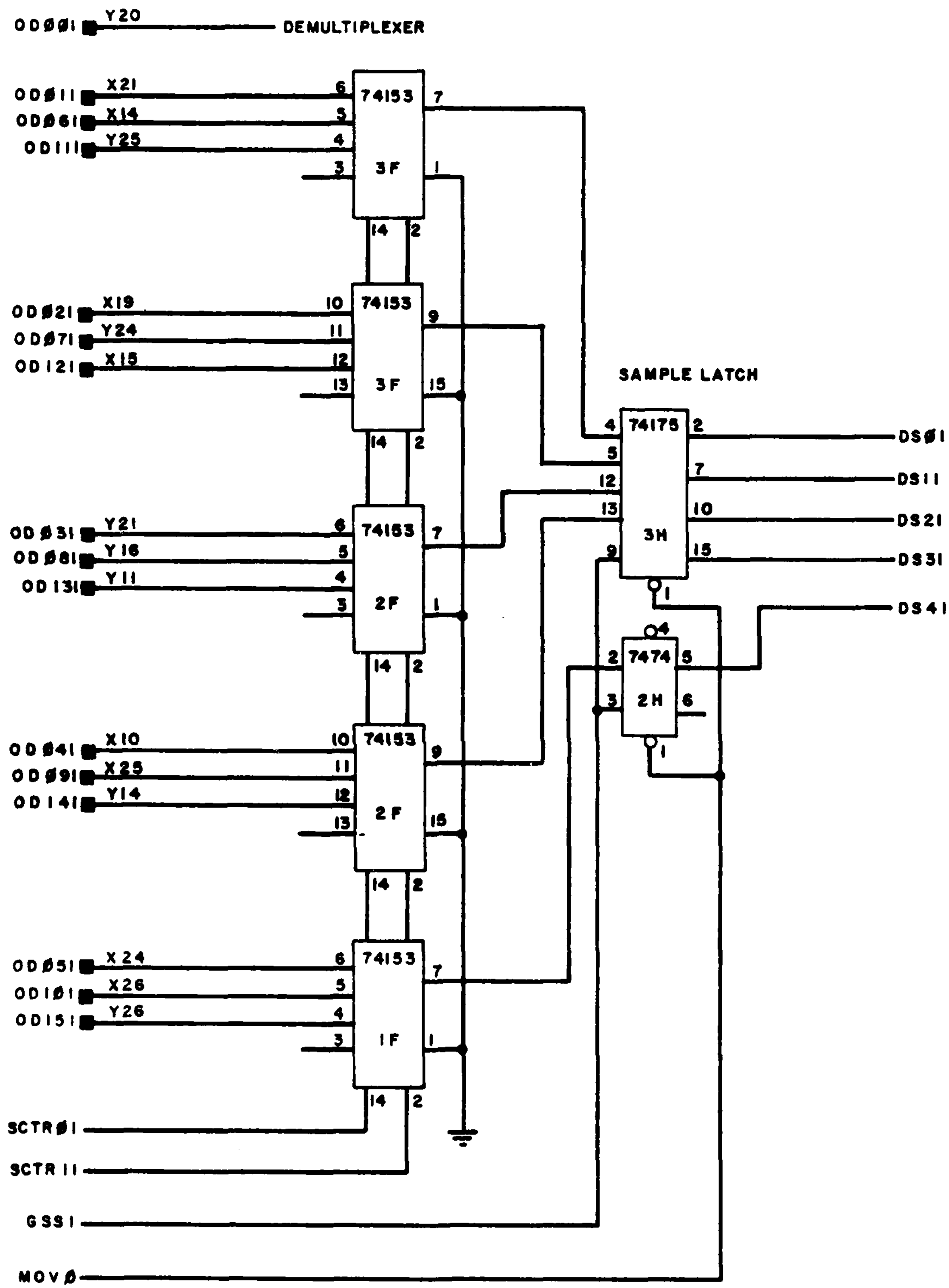
Bit 4 (12), CM041, is the start bit. This bit clears the telephone voice decoder logic and places the decoder in its initial state so that a voice signal will be generated as soon as voice data are transferred from the processor.

The final command bit (CM051) is the pickup bit. This signal activates a relay which is connected to the telephone interface. When this bit is active, the relay contact closes and thus "picks up" the line. This act is synonymous with picking up the receiver of a telephone instrument. Two-way voice communication may proceed after the pickup operation is initiated. When the pickup bit is inactive, the relay contact opens and the line is hung up. This is synonymous with hanging up the telephone receiver of a telephone instrument.

	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
N				PCB77											
M	RELAY	7400	UAF41	UAF41			7403	7403	21L02	21L02	21L02	7408	7402	7400	7400
L	RELAY	UAF41	FILTER PARTS	UAF41			7403	7403	21L02	21L02	21L02	74123	7486	7400	7474
K	74107	74107	7408	OP-01			7403	74175	21L02	21L02	21L02	ONE SHOT PARTS	7404	7408	7474
J	7402	7404	7408	DAC76			7408	74175	21L02	21L02	21L02	74123	7402	7408	7474
H	7474	7474	74175	PARTS			7400	74175	21L02	21L02	74157	74123	74197	74197	74197
F	74153	74153	74153	74123			7474	74175	21L02	21L02	74157	74157	74197	74197	74197

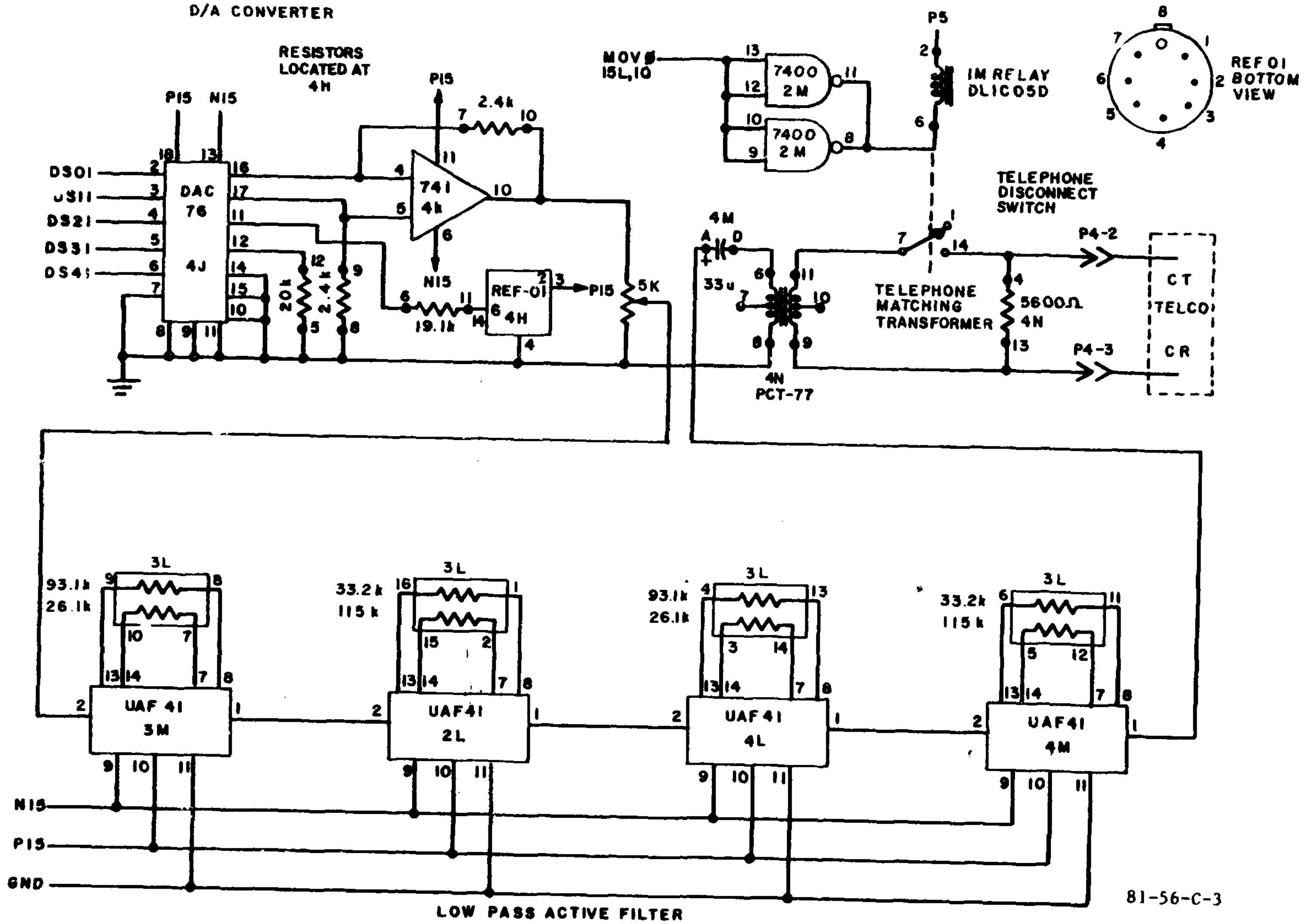
81-56-C-1

FIGURE C-1. MDB BOARD STUFFING CHART (VOICE OUTPUT VIA TELEPHONE WITH MEMORY)



81-56-C-2

FIGURE C-2. DEMULTIPLEXOR LOGIC



81-56-C-3

FIGURE C-3. ANALOG CIRCUITS

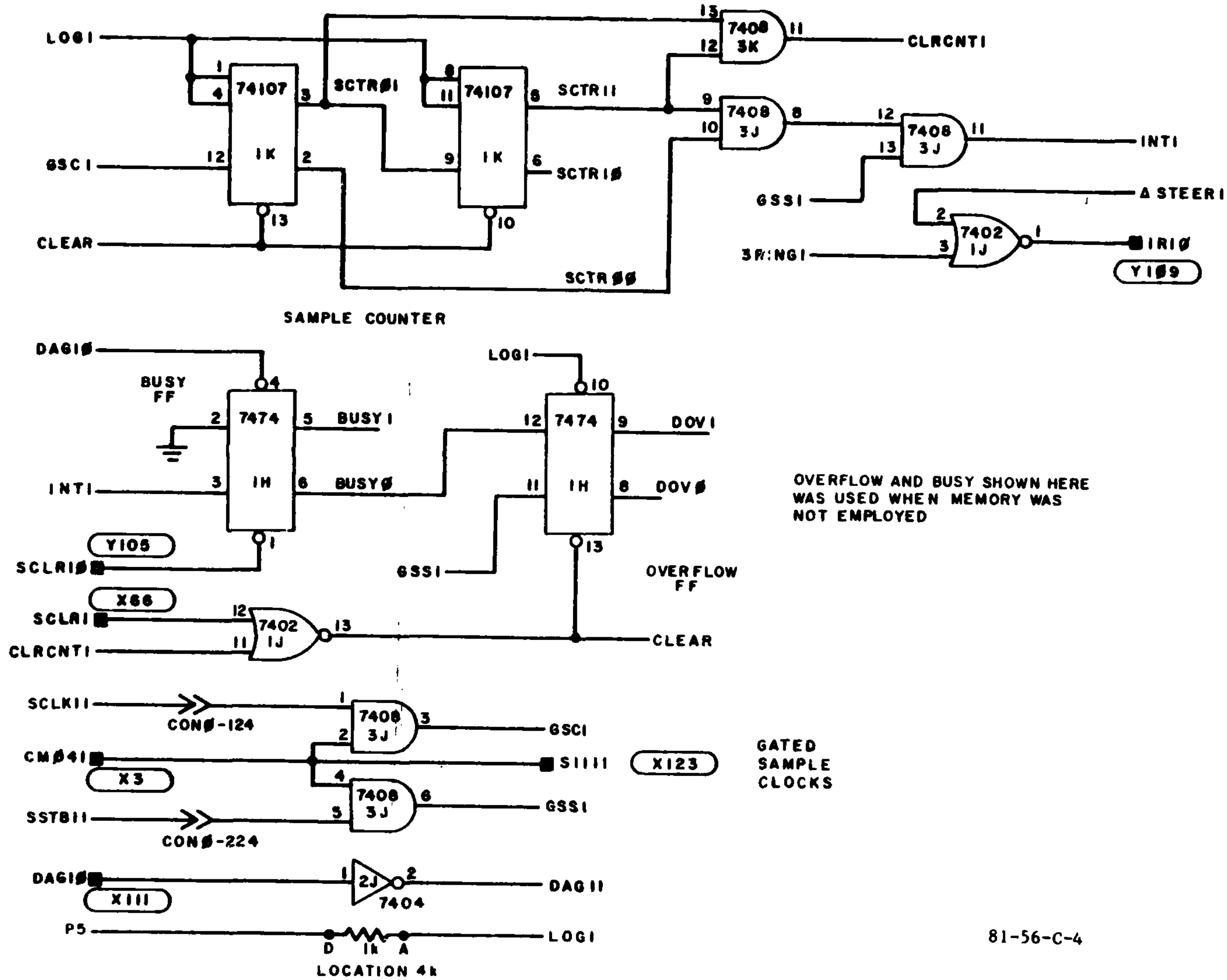
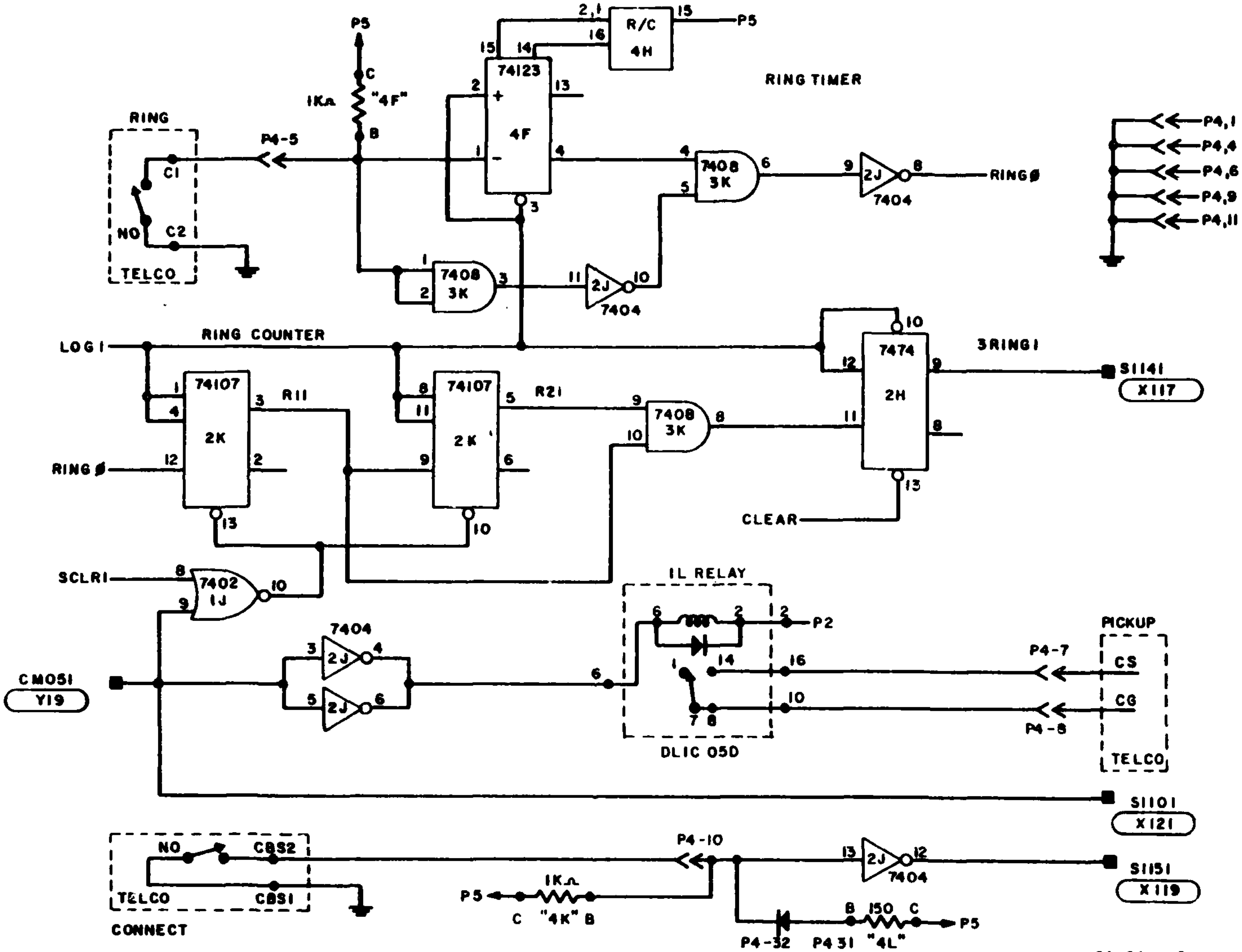


FIGURE C-4. CONTROL LOGIC, TELEPHONE



81-56-C-5

FIGURE C-5. TELEPHONE LOGIC

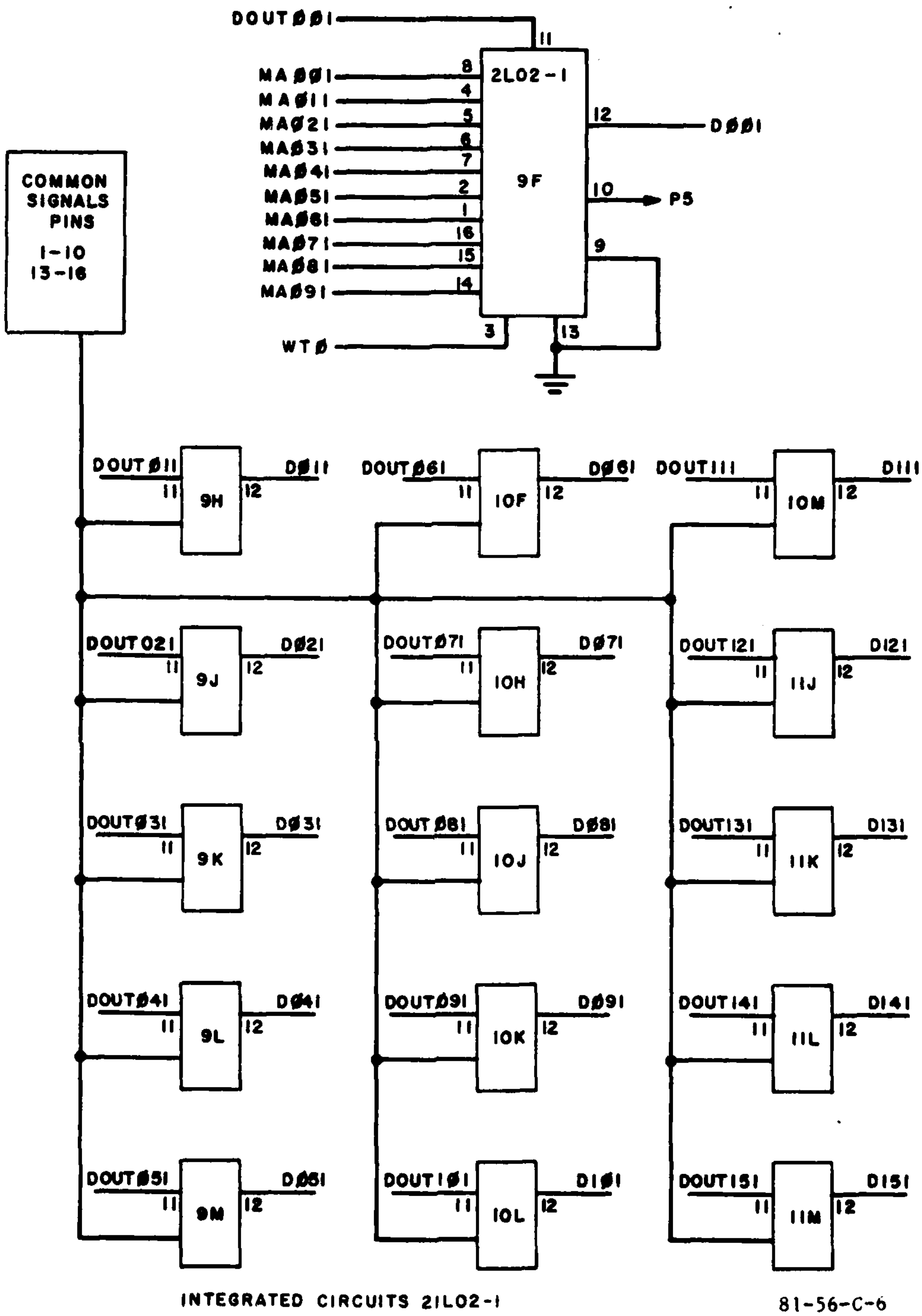
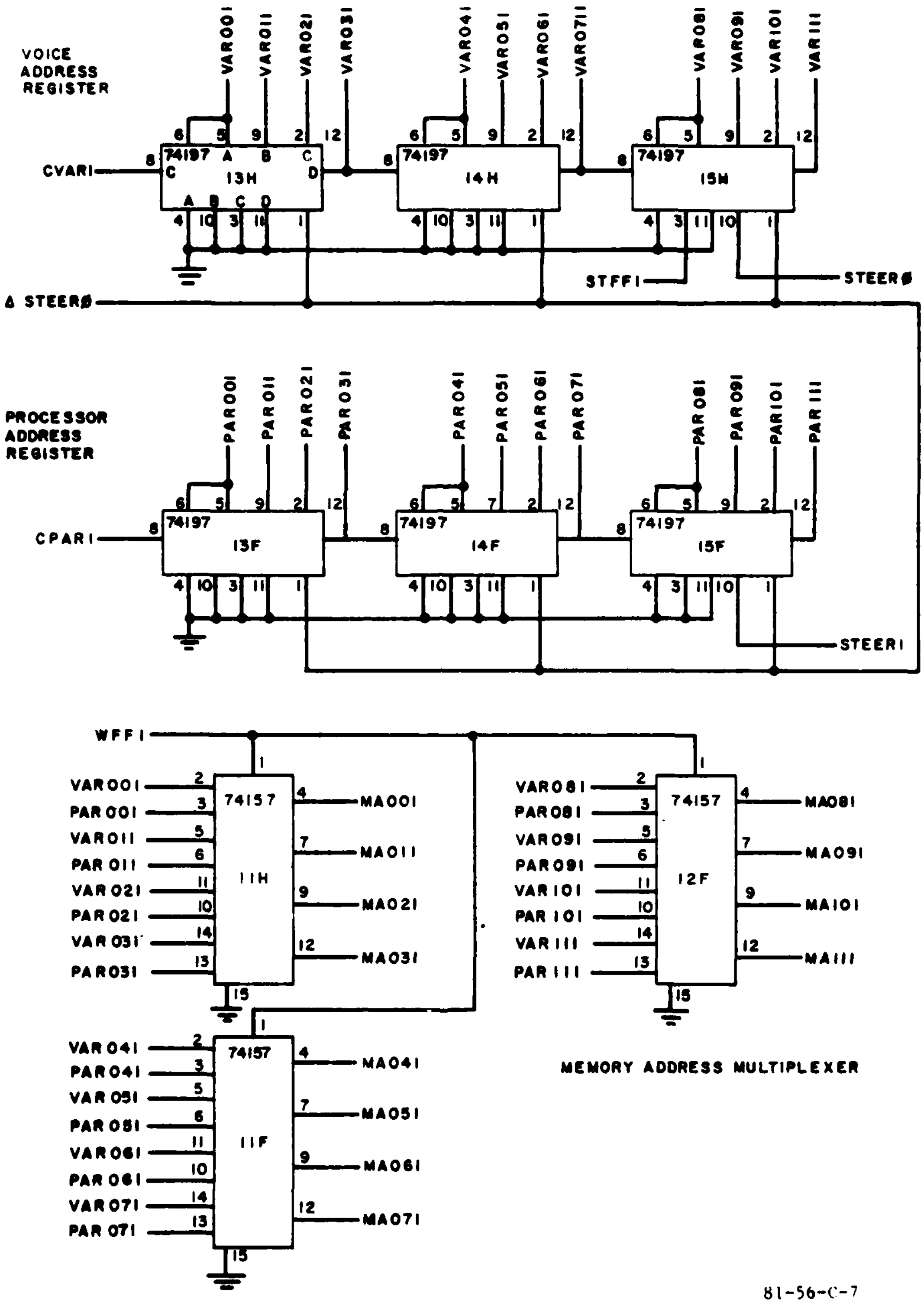
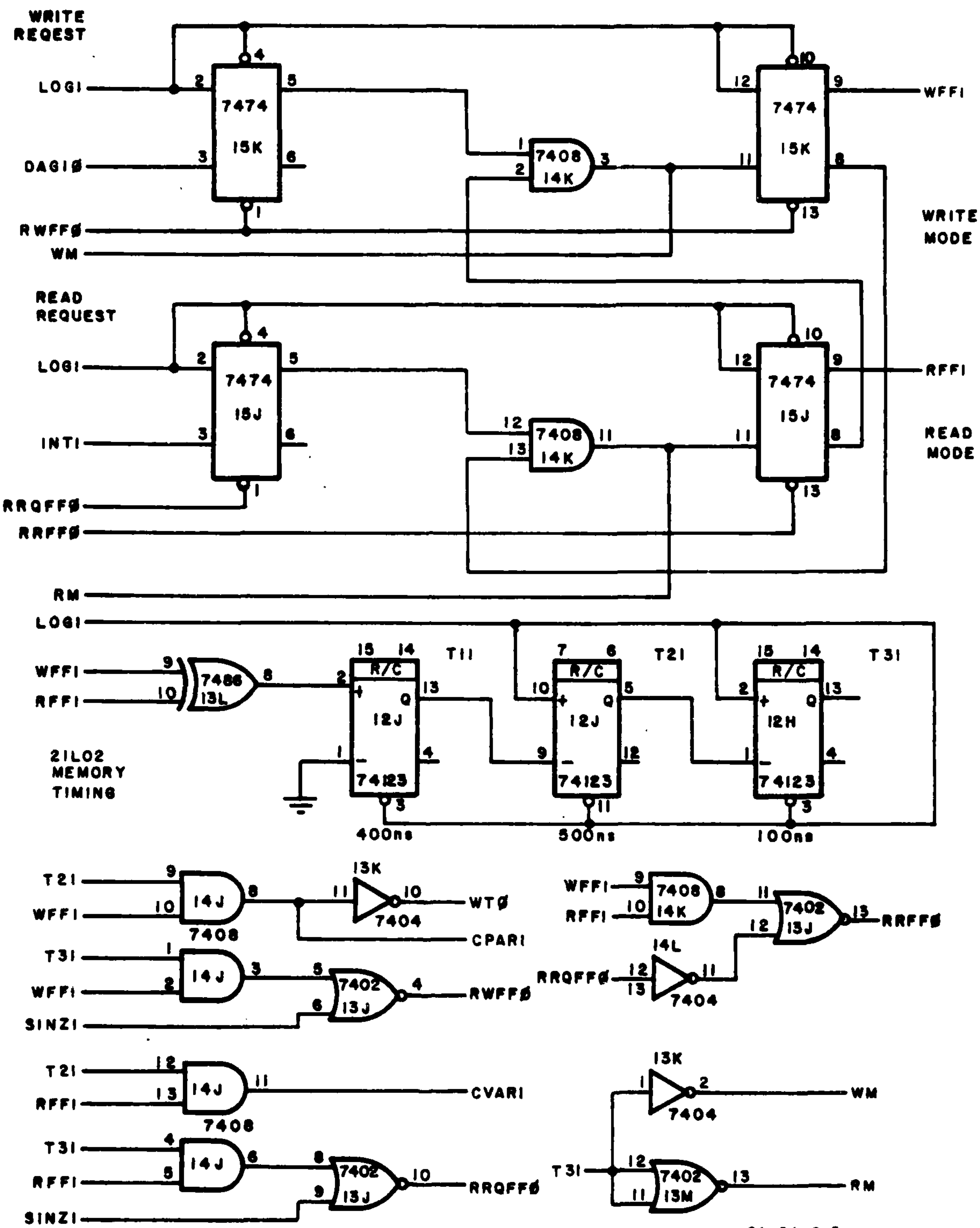


FIGURE C-6. MEMORY ELEMENTS



81-56-C-7

FIGURE C-7. MEMORY ADDRESS COUNTERS AND MULTIPLEXOR



81-56-C-8

FIGURE C-8. MEMORY CONTENTION LOGIC

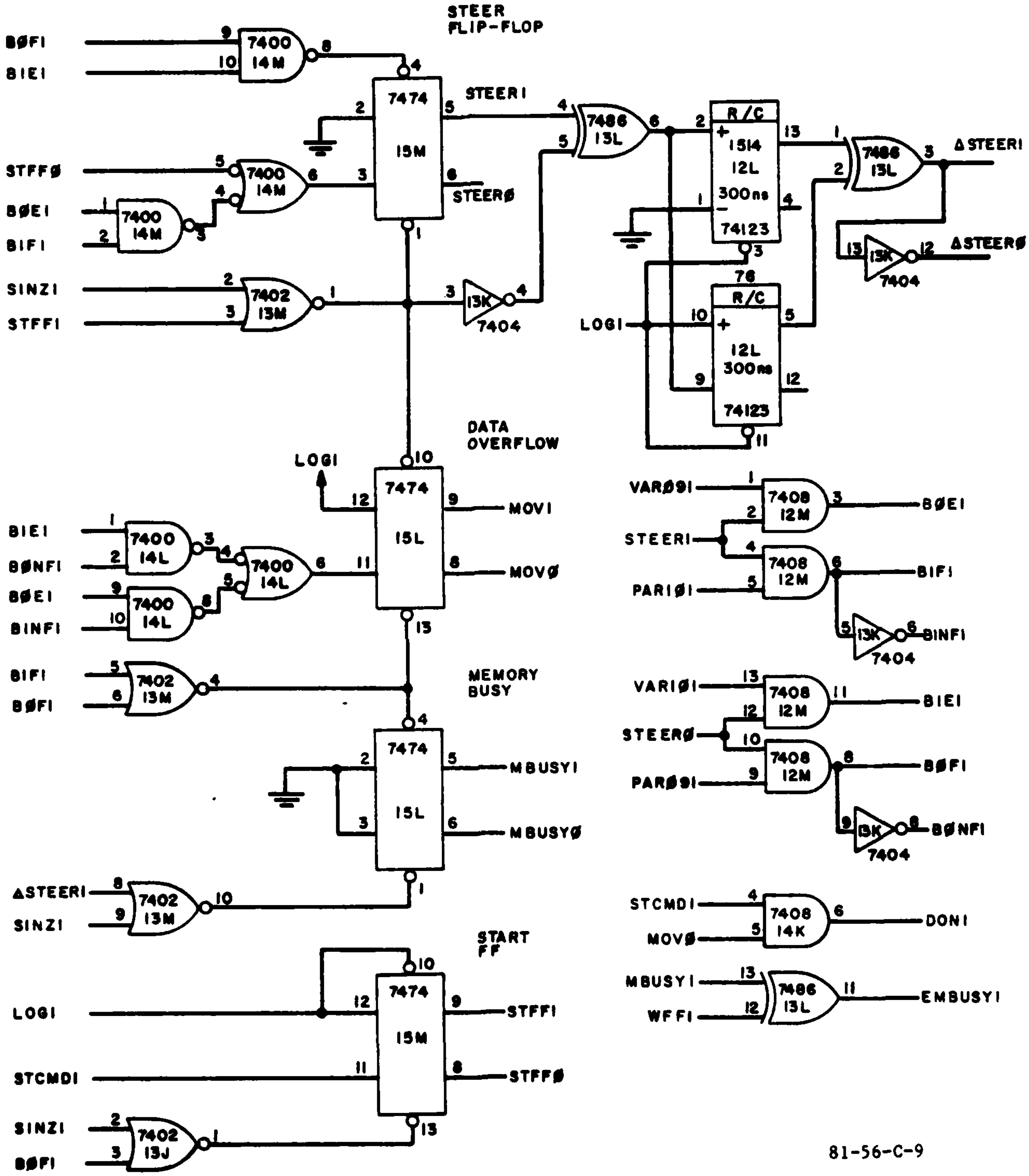
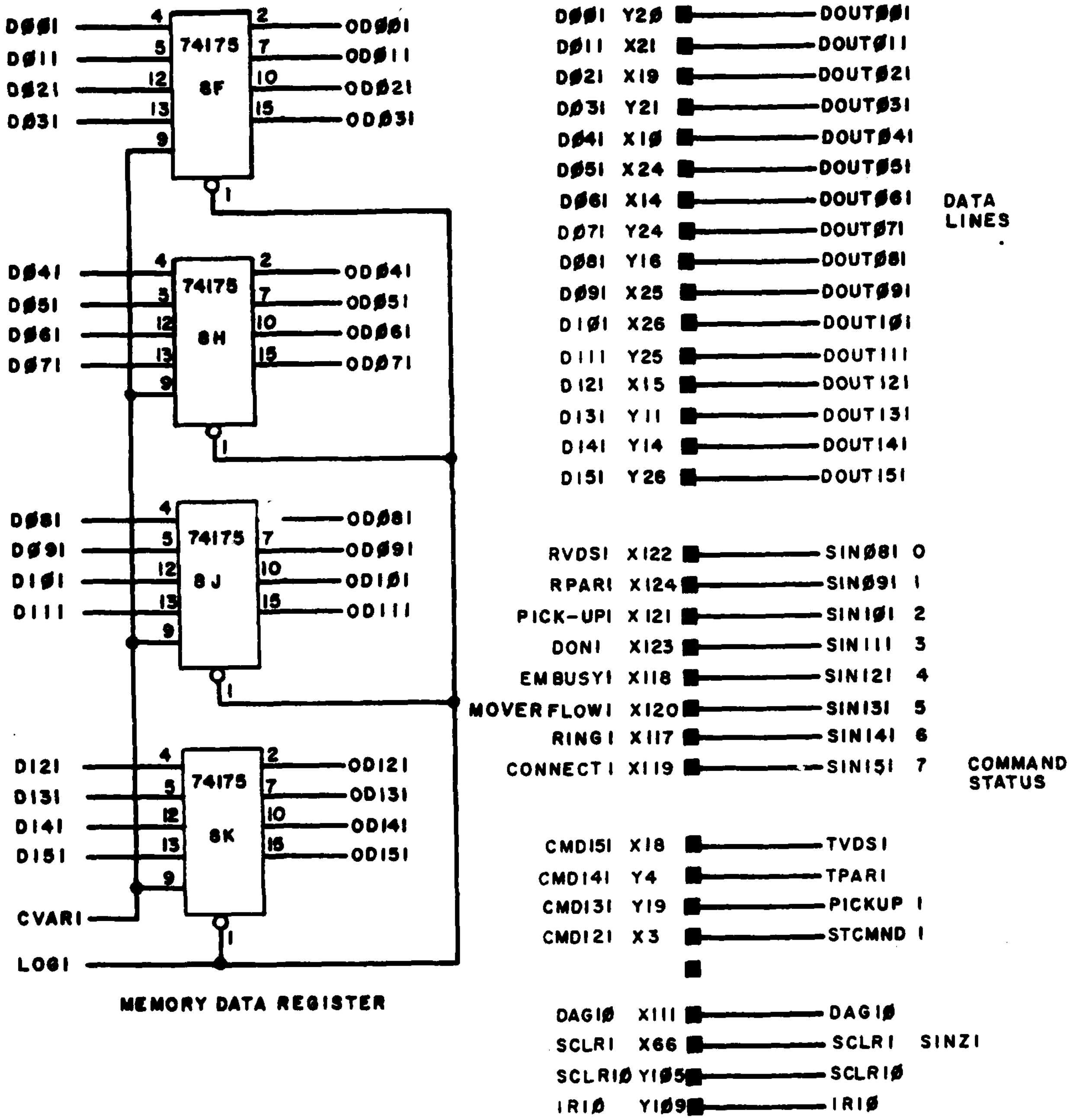


FIGURE C-9. MEMORY CONTROL LOGIC



81-56-C-10

FIGURE C-10. MEMORY DATA REGISTER AND COMPUTER BUS SIGNALS

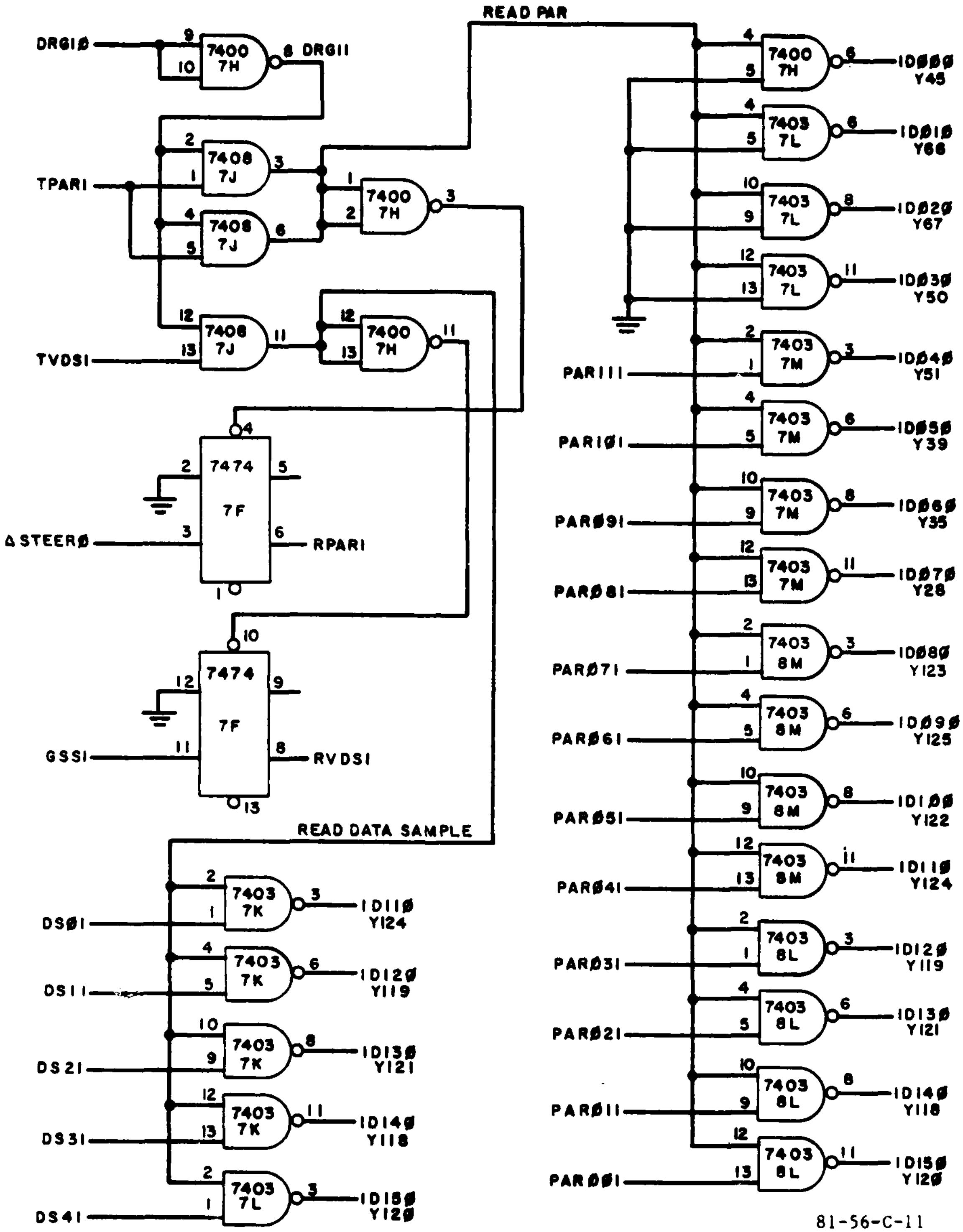


FIGURE C-11. TEST LOGIC

APPENDIX D

TELEPHONE LINE CONTROLLER/CROSSPOINT SWITCH

PREFACE.

All of the telephone line switching within the Mass Weather Dissemination System is performed by the crosspoint switch. The crosspoint switch can connect any of the 20 incoming telephone lines to either of the two fast-file flight plan recorders or to any one of the four flight service specialist positions. Control of the crosspoint switch is achieved via data and commands which are received from the 7/32 processor. The crosspoint switch logic diagram is also included in this appendix.

CROSSPOINT SWITCH LOGIC.

The heart of the crosspoint switch is a Clare 969-A48-A2E Mini Memory Matrix Module. Each of these modules is a self-latching 8 by 8 crosspoint switch. Control of this module is provided via 16 data bits and a strobe pulse. Eight of the data bits represent eight of the incoming telephone lines while the remaining eight data lines represent system options, such as fast-file flight plan recorders and specialist positions. After the 16 data bits have been established, a strobe pulse is sent to the module which actually performs the switching function. The module holds this connection until instructed to change it.

Three Clare modules were interconnected to form a 24 by 8 crosspoint switch. Of this capacity, 20 by 6 was actually utilized by the Mass Weather Dissemination System. The eight X control lines of the three Clare modules are parallel connected, as are the Y control lines. The Y audio lines of the three modules are also parallel connected. The Y audio lines Y6, Y1, Y2, Y3, Y4, and Y5 are connected to system options SPECIALIST1, SPECIALIST2, SPECIALIST3, SPECIALIST4, FASTFILE1, and FASTFILE2, respectively. Telephone lines 0 through 7 are connected to Clare module one's X audio lines X0 through X7, respectively. Similarly, telephone lines 8 through 15 are connected to Clare module two's X audio lines X0 through X7, and telephone lines 16 through 19 are connected to Clare module three's X audio lines X0 through X3.

Since the X control and Y control lines of the three Clare modules are parallel connected, the strobe pulse must be directed to only one of the modules. This is accomplished by having three separate commands. Commands 01, 02, and 03 activate Clare modules 1, 2, and 3, respectively.

CROSSPOINT SWITCH OPERATION.

The crosspoint switch is utilized when the 7/32 processor determines that a pilot on one of the incoming telephone lines has requested to be connected to a system option. The 7/32 determines which of the system options are available for use, so that it can generate the proper 16-bit control word, as well as the proper command for the crosspoint controller. The 7/32 processor then sends the 16-bit control word to the crosspoint controller, followed immediately by the command. Approximately 10 milliseconds after the crosspoint controller receives the command, the crosspoint connection is completed and the pilot has been connected to the system option of his choice.

THE CROSSPOINT CONTROLLER.

The crosspoint controller is fabricated on an MDB Universal Interface Logic Module number MDB-48-013-16-01. This module contains all of the logic required for communication of data on the Interdata 7/32 processor I/O bus and provides space for custom logic.

The cross nt controller has a 16-bit data register to hold the 16-bit crosspoint control word. Bits OD001 through OD071 represent the Y control lines Y7 through Y0, respectively, and bits OD081 through OD151 represent the X control lines X7 through X0, respectively. Since the Clare module requires 48 volts on the control lines, each data bit from the crosspoint controller activates a reed relay (DL-1C-05D) which converts the standard 5-volt logic to 48 volts.

Immediately after the 16-bit word has been sent to the crosspoint controller, the strobe command is sent to the crosspoint controller. The strobe command initiates a timing chain of three one-shots and also informs the 7/32 processor that the crosspoint controller is not available to receive switching commands via the busy bit (BUSY1 is high). The first one-shot is a delay of 7 milliseconds which allows the reed relays time to settle. When the 7 millisecond delay is complete, the second one-shot fires for one millisecond, and this becomes the strobe pulse. At the end of the strobe pulse, the third one-shot fires and clears out the command and crosspoint control word registers. The clear pulse from the third one-shot also clears out the busy flag to the 7/32 processor indicating that the crosspoint controller is available for more switching commands.

A feature of the Mass Weather Dissemination System that is associated with the crosspoint switch is the audio test panel. This panel allows any one of the 20 telephone lines to be monitored for test and demonstration purposes. The audio test panel consists of 20 audio jacks, 20 light emitting diode (LED) indicators, and an isolating audio amplifier.

The LED indicator, which is located above the audio jack, lights up whenever that telephone line is in use.

To monitor a particular telephone line, the operator plugs the patch-cord plug into the appropriate telephone line jack. The audio amplifier amplifies the signal and plays it out over a loudspeaker.

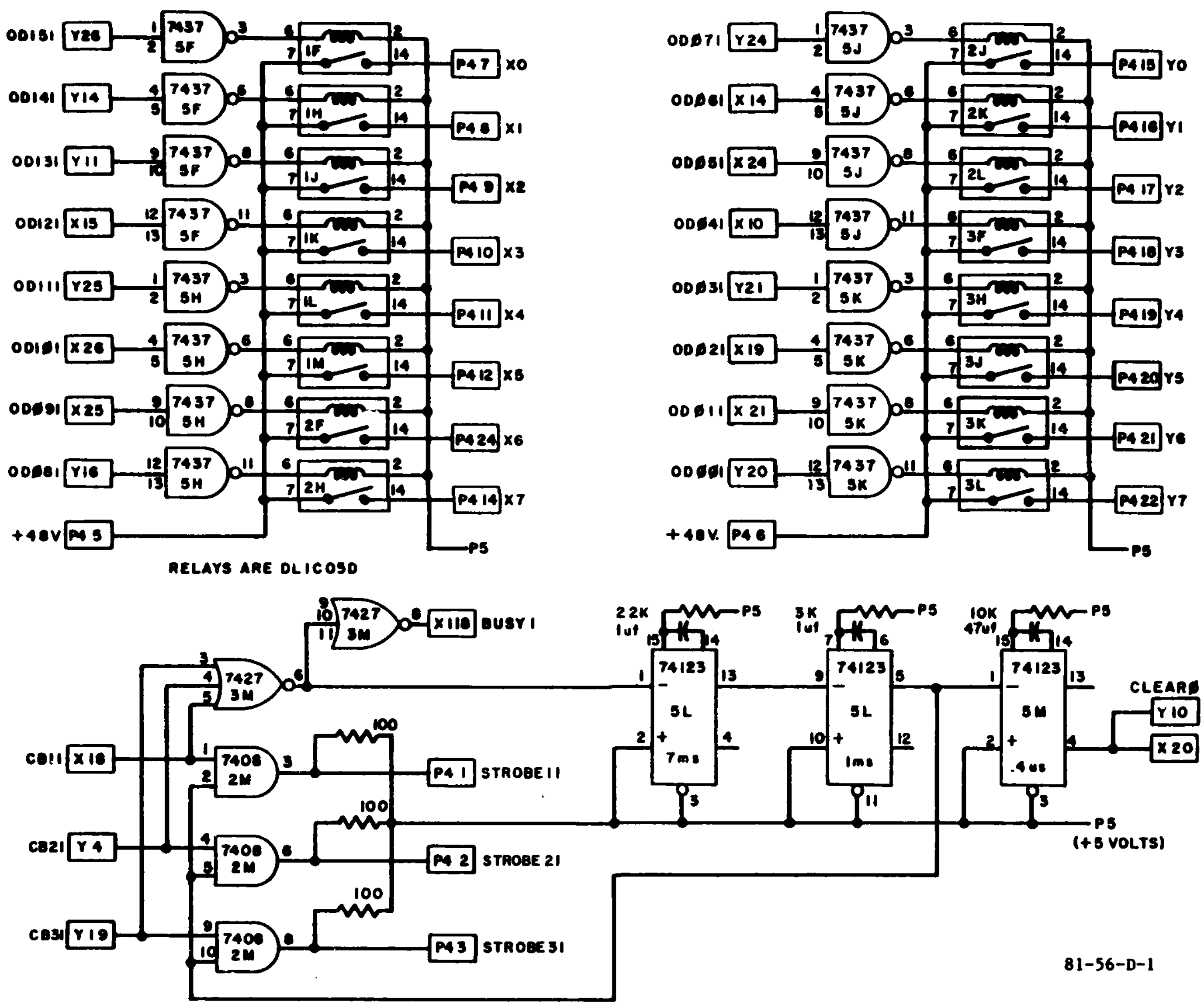


FIGURE D-1. CROSSPOINT SWITCH CONTROL

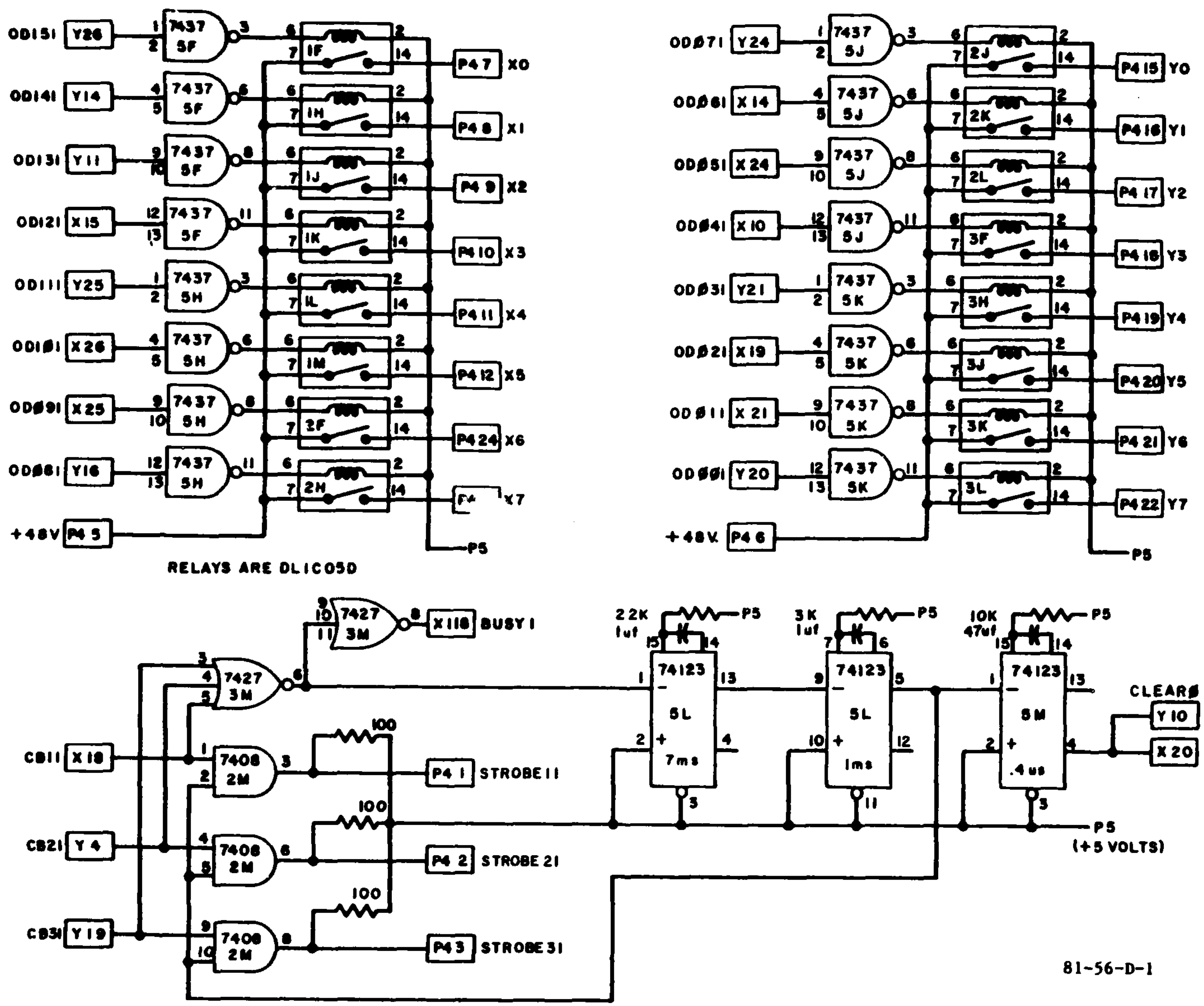


FIGURE D-1. CROSSPOINT SWITCH CONTROL

APPENDIX E

SPECIALIST POSITIONS

PREFACE.

The Mass Weather Dissemination System Exploratory Engineering Model incorporates four specialist positions. These positions consist of four standard telephone instruments which were modified to be compatible with the Mass Weather Dissemination System. These telephone instruments can be connected to any of the 20 incoming telephone lines and are used whenever a pilot requests to speak to a flight service specialist or when the system determines that the utterance recognition device (URD) is unable to identify a pilot's request.

THE SPECIALIST INTERFACE.

The specialist positions are controlled asynchronously by the 7/32 processor. Audio connection to the pilot is accomplished via the crosspoint switch array.

The control interface is designed around a General Instruments AY-5-1013 universal asynchronous receiver transmitter (UART). Communications between the computer and the interface are in accordance with Electronics Industry Association (EIA) standard RS-232C. The format is 1200 baud, 8 data bits, 2 stop bits, and no parity.

The interface is designed in such a fashion as to emulate a remote terminal. This allows the programmable asynchronous line modules (PALM) to control the specialist positions without extra driver software development. This also allows the specialist positions to be located remote from the 7/32 processor without extensive buffering of data lines.

The 7/32 processor sends only one, single byte, command to the interface. This is the ring command, X'01'. The reception of this command causes the UART signal DAV1 to go high, firing a one-shot of duration 20 milliseconds. During this period, DG1 is high and is logically "nanded" with the least significant bit (LSB) of the received byte. The resulting low going pulse sets two D flip-flops. The signal, BE1, of one flip-flop goes high, sounding the buzzer circuit on the appropriate specialist phone. The second flip-flop's signal, PUE1, goes high, allowing the interface to acknowledge that a specialist has picked up his instrument. During the period of DG1, the complimentary signal R10 goes low, inhibiting any transients which might follow the command byte.

The specialist phone instruments are equipped with two status switches. The on-line and off-line switch is located on the front of the phone, and its state is indicated by a light emitting diode (LED). The debounced state of this switch is connected to bit 1 of the UART's transmitter. The second status switch is the pickup/hangup switch which is located in the cradle of the phone. The debounced output of this switch is connected to the LSB of the UART's transmitter input. The status codes returned to the 7/32 computer by the interface are off-line hung up (X'40'), off-line picked up (X'41'), on-line hung up (X'42'), and on-line picked up (X'43').

AD-A125 655 HARDWARE DESCRIPTION OF MASS WEATHER DISSEMINATION SYSTEM EXPLORATORY ENG..(U) FEDERAL AVIATION ADMINISTRATION TECHNICAL CENTER ATLANTIC CIT.. P QUICK

2/2

UNCLASSIFIED SEP 82 DOT/FAA/CT-81/56 DOT/FAA/RD-82/32 F/G 9/2 NL

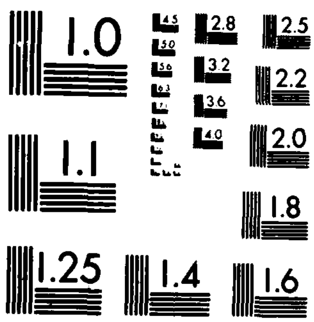
END

DATE FILMED

4 83

DTIC

M-2



MICROCOPY RESOLUTION TEST CHART
NATIONAL BUREAU OF STANDARDS-1963-A

A state transition on any of the aforementioned switches causes a one-shot to be triggered. This results in a transmit strobe, TSO, to be presented to the appropriate UART. In the case of pickup, the buzzer enable signal, BE1, is turned off, thereby, inhibiting the phone's buzzer. During the transmission of a status byte to the 7/32 processor, the serial input to the UART is disabled so as to inhibit the data wraparound feature of the PALM. The wraparound is usually used to display a character on a remote terminal.

The buzzers in the phone units are 8 OHM speakers driven through an emitter follower transistor (TIP-29) by an astable oscillator having an approximate frequency of 138 hertz. This buzzer may be inhibited by a switch located on the specialist phone. A blinking led on the phone is activated at the same time as the buzzer. The led is driven by the same astable as the buzzer, but the frequency is divided by 128 to reduce the blink rate to the visible range.

THE TELEPHONE INSTRUMENT.

The specialist phone units are modified Stomberg-Carlson 500D/554B telephones. The ringer and dial components of the phones have been removed. Jumpers were installed at various points to complete the audio path to the handset.

Incoming phone lines to the Mass Weather Dissemination System are obtained through a CDH interface. This provides the bidirectional pair customer-tip (CT) and customer-ring (CR). Connection to the specialist phones is through a balanced transformer to provide added isolation. The CT-CR pair does not provide a talk battery voltage, necessitating that such a voltage be supplied at the isolation transformer.

Three possible talk voltages are available. The normal operational voltage is approximately 1 volt. The second voltage of 5 volts is used during demonstrations because it provides a higher level of audio to an external speaker; unfortunately, this voltage generates a hiss on the specialist instrument. The third voltage is externally applied and is used to provide a variable talk bias for in-house testing. These voltages are switch-selectable on the isolation board.

HARDWARE CONSIDERATIONS.

The interfaces controlling the four specialist phones are located on a single card located in the mass dissemination processor bay. Power supply requirements are +5 volts, +15 volts, and -12 volts. The +5 volt supply is generated by the same lambda power supply that is used to power the voice boards. The +15 and -15 volts are supplied by the Interdata processor and are reduced to the proper levels for the interface by a zener diode arrangement. A 47UF capacitor is provided for each row of integrated circuits on the interface card to provide decoupling.

The signal description notation used in this report is the same as found in the Interdata documentation. The least significant number in a signal's name specifies the active state. The other digits in a signal's name refer to the channel involved. For example, the signal DAV31 represents data available when in the logic-one state for specialist position three.

MNEMONICS

BE1	Buzzer enable
CCO	Condition change switch transition generates status byte
CR	Customer ring — Telco signal
CT	Customer tip — Telco signal
DAV1	Data available — indicates a command byte has been received
DG1	Data gate — enables ring and picku,
EOC	End of character — low during transmit
GSIN	Gated serial input — TTL level
INITO	Initialize pulse — transmits status on power-up
LOG'1'	Logic one — +5 volts through a 1 KOHM resistor
PUE1	Pickup enable — allows pickup to trigger one-shot
RDAVO	Reset data available
RIO	Receive inhibit
RING1	Enables buzzer at DG1 time
SIN	Serial input — EIA RS-232C
SOUT	Serial output — EIA RS-232C
TSO	Transmitter strobe

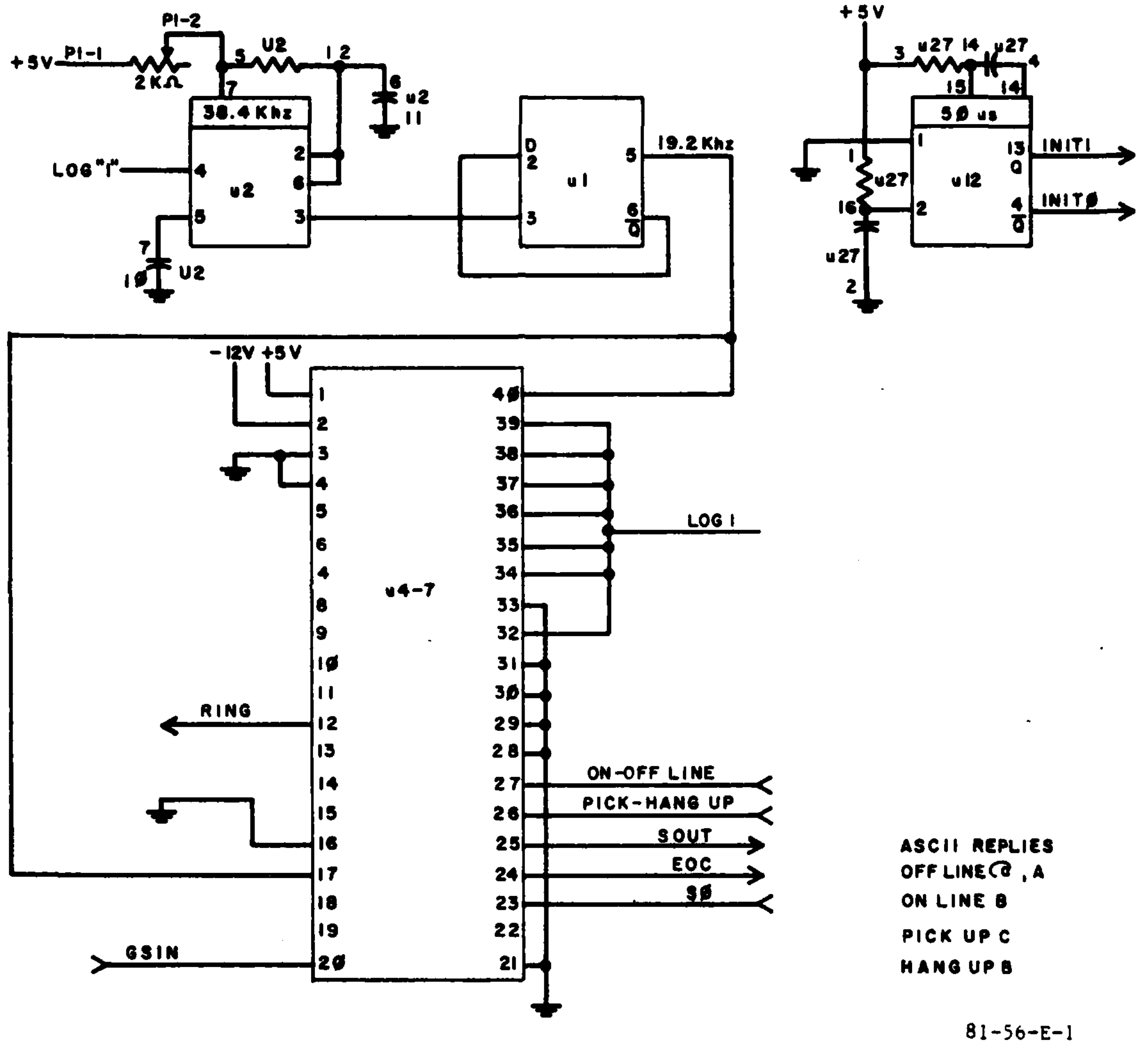
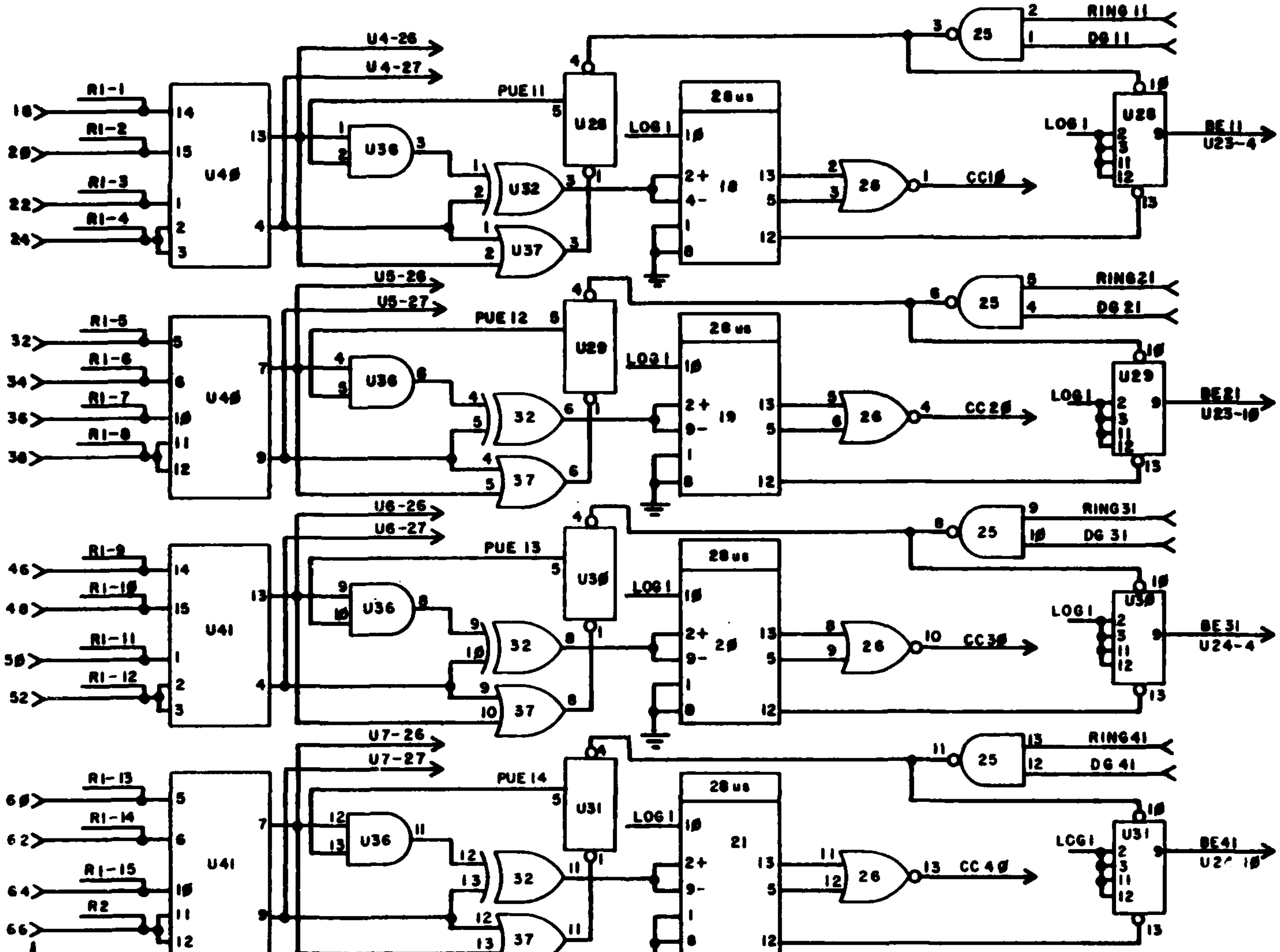
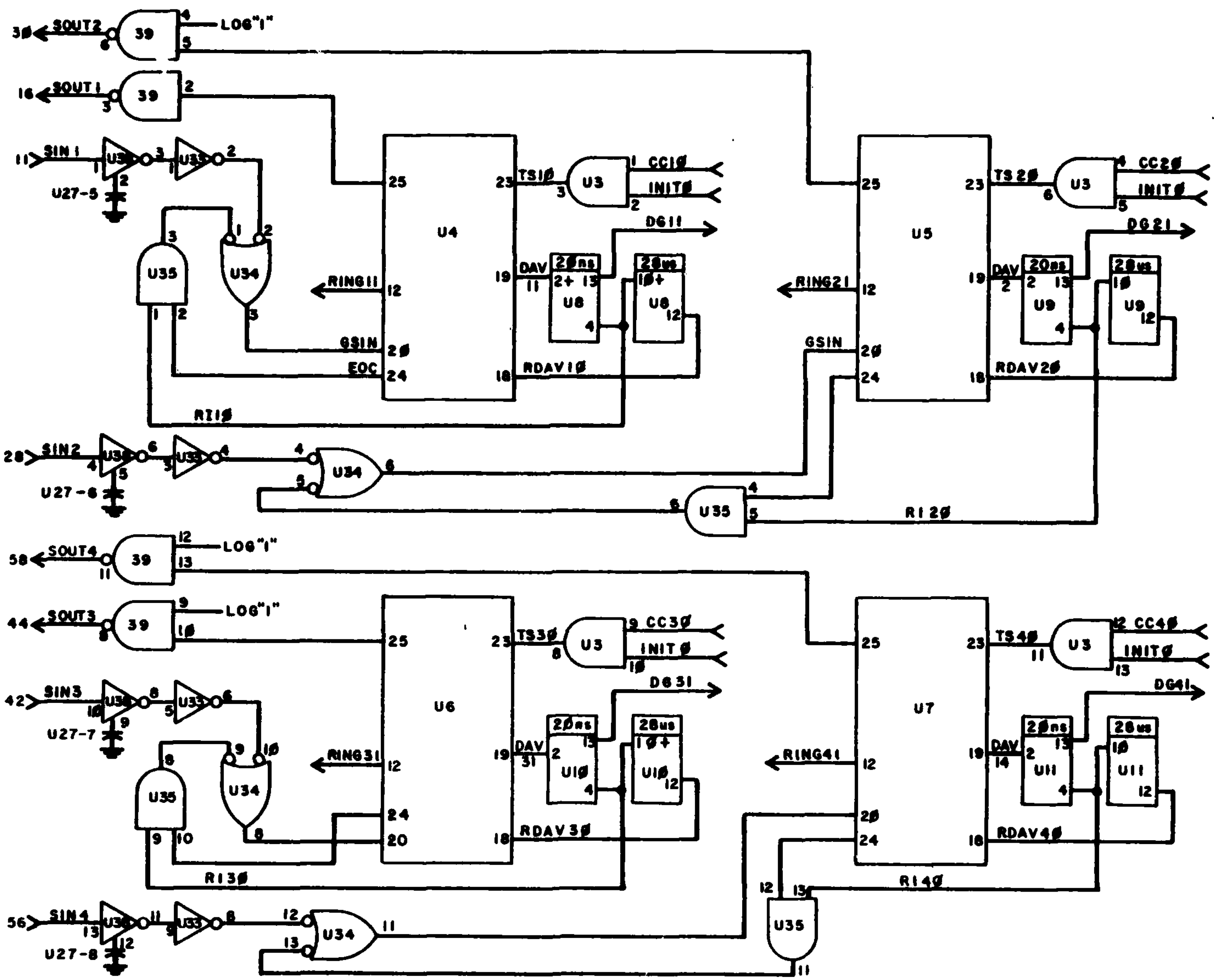


FIGURE E-1. SPECIALIST INTERFACE



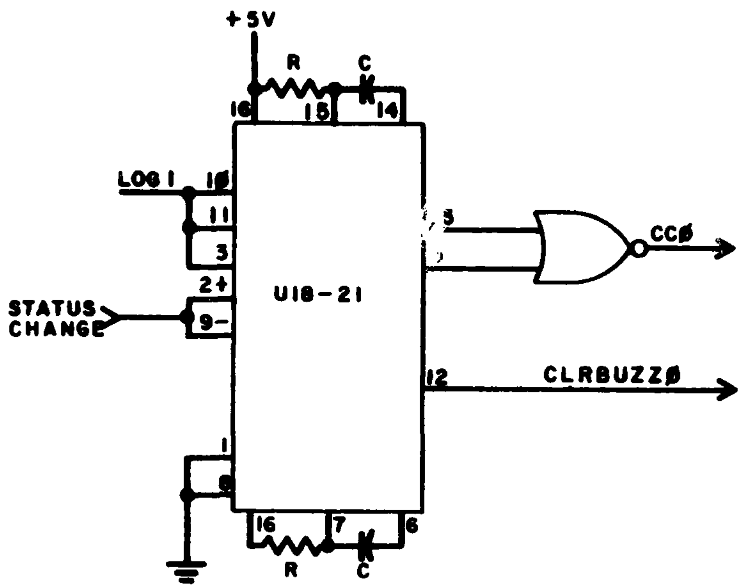
81-56-E-2

FIGURE E-2. FOUR-SPECIALIST INTERFACE (SHEET 1 OF 2)

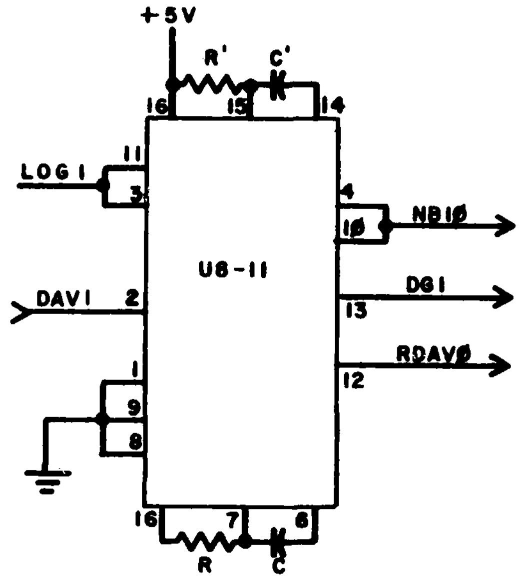


81-56-E-2

FIGURE E-2. FOUR-SPECIALIST INTERFACE (SHEET 2 OF 2)



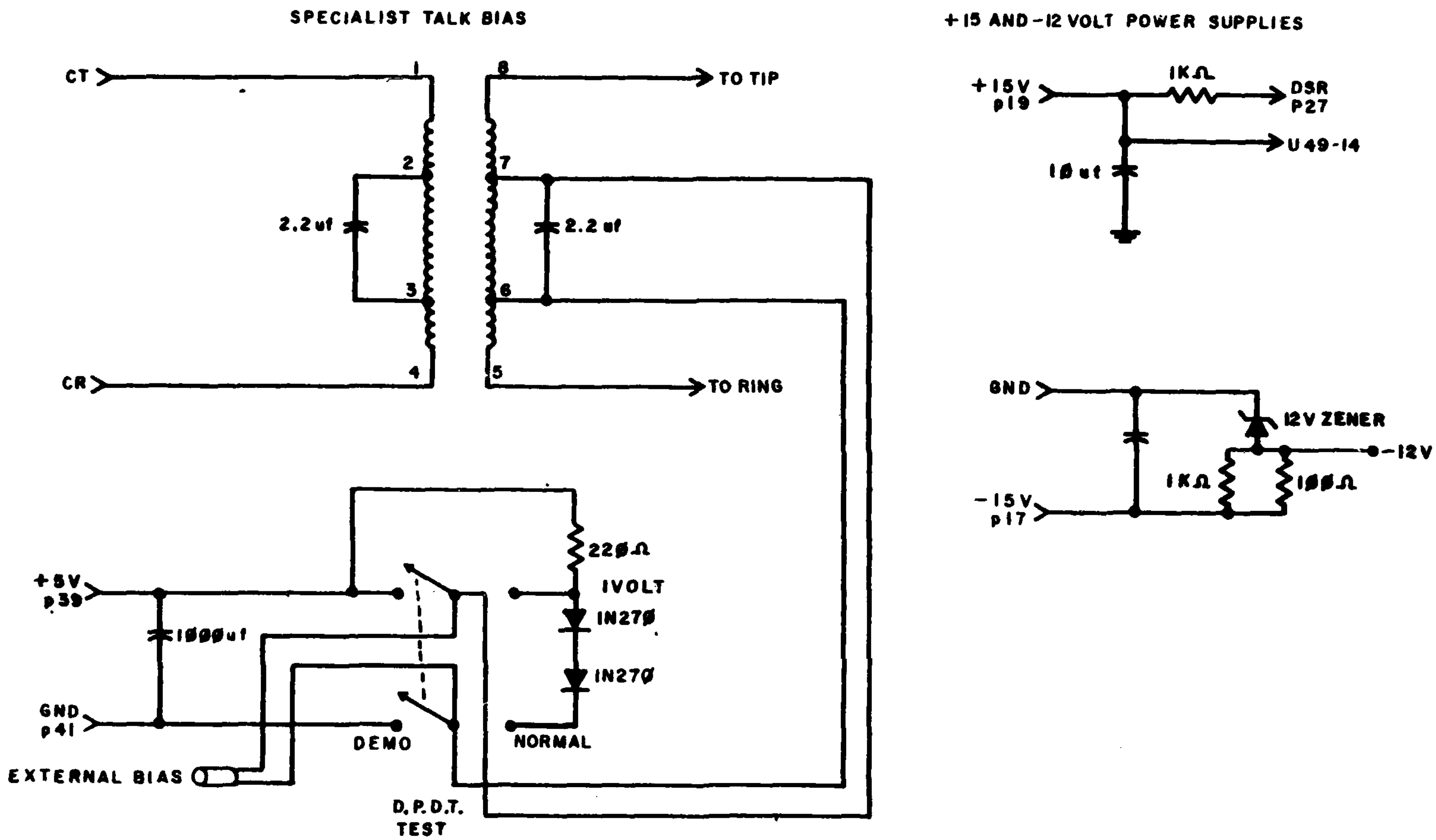
THE VALUES OF RC TIME CONSTANTS ARE RESPECTIVELY 1ØK AND .Ø1uf. THESE BRIDGES ARE LOCATED ON U16 AND U17. ON BOTH PACKAGES OF BRIDGES PINS 1, 3, 5 AND 7 ARE CONNECTED TO +5V. CONNECTION OF THE BRIDGES TO U18 – 21 AS INDICATED ABOVE ARE MADE SEQUENTIALLY. EG. U18 USE THE FIRST 2 BRIDGES ON PACKAGE U16. U19 USES THE NEXT 2 ETC. THESE ONE SHOTS HAVE A PULSE DURATION OF 28usec.



THE VALUE OF THE R'C' TIME CONSTANTS ARE RESPECTIVELY, 1ØK AND 6.8uf. THESE ARE THE 4 BRIDGES LOCATED ON U14 THE RC BRIDGES ARE LOCATE ON U15 HAVING VALUES OF 1ØK AND Ø1uf. ON BOTH U14 AND U15 PINS 1, 3, 5 AND 7 ARE TIED TO +5V. THE RESPECTIVE CONNECTIONS ARE MADE SEQUENTIAL R'C' GENERATES A PULSE DURATION OF 2Ø m sec. RC RESULTS IN A PULSE OF 28usec.

81-56-E-3

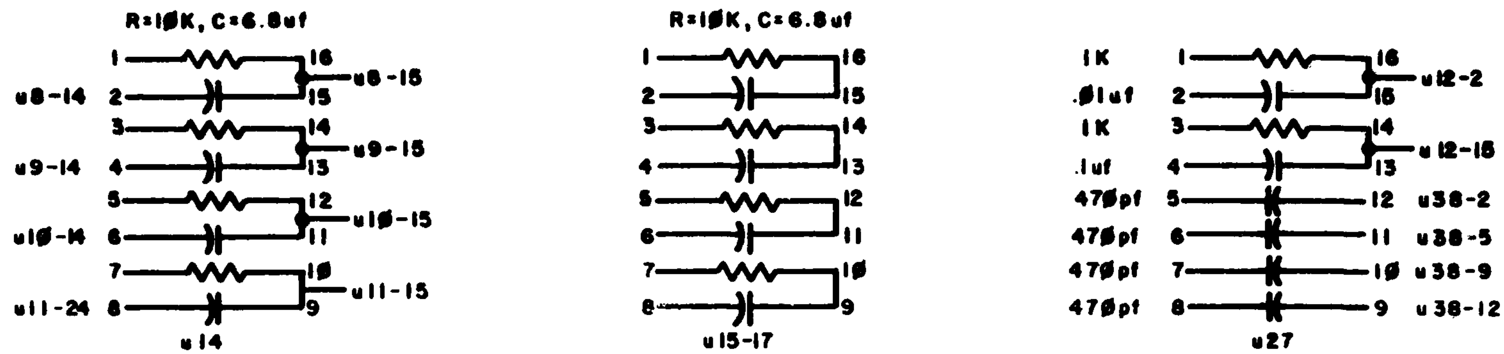
FIGURE E-3. FOUR-SPECIALIST INTERFACE, TIMING CONNECTIONS

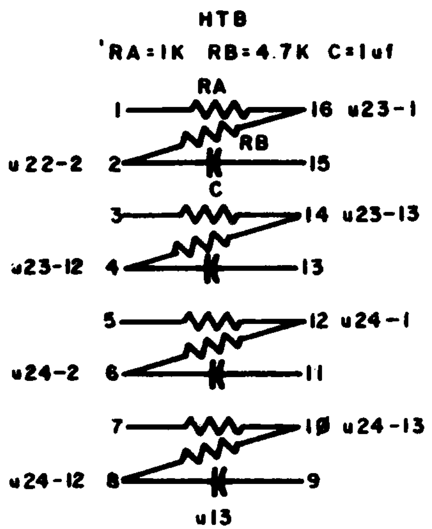


SWITCH POSITIONS
NORMAL-OPERATIONAL LEVEL
DEMO-FOR DRIVING EXTERNAL SPEAKER REDUCED VOICE QUALITY.
TEST-USED FOR URD EVALUATIONS IN HOUSE.

81-56-1-4

FIGURE E-4. SPECIALIST INTERFACE, SPECIALIST TALK BIAS





NOTE

u1-7474	u22-NOT USED
u2-NE555	u23-NE556
u3-74Ø8	u24-NE556
u4-AY-5-1Ø13	u25-74ØØ
u5-AY-5-1Ø13	u26-74Ø2
u6-AY-5-1Ø13	u27-TIMING
u7-AY-5-1Ø13	u28-7474
u8-74123	u29-7474
u9-74123	u3Ø-7474
u1Ø-74123	u31-7474
u11-74123	u32-7486
u12-74123	u33-74Ø4
u13-TIMING	u34-74ØØ
u14-TIMING	u35-74Ø8
u15-TIMING	u36-74Ø8
u16-TIMING	u37-7432
u17-TIMING	u38-LM1489
u18-74123	u39-LM1488
u19-74123	u4Ø-74279
u2Ø-74123	u41-74279
u21-74123	u42-1K RESISTOR ARRAY

81-56-E-5

FIGURE E-5. DISCRETE COMPONENT, INTEGRATED CIRCUIT PART NUMBERS

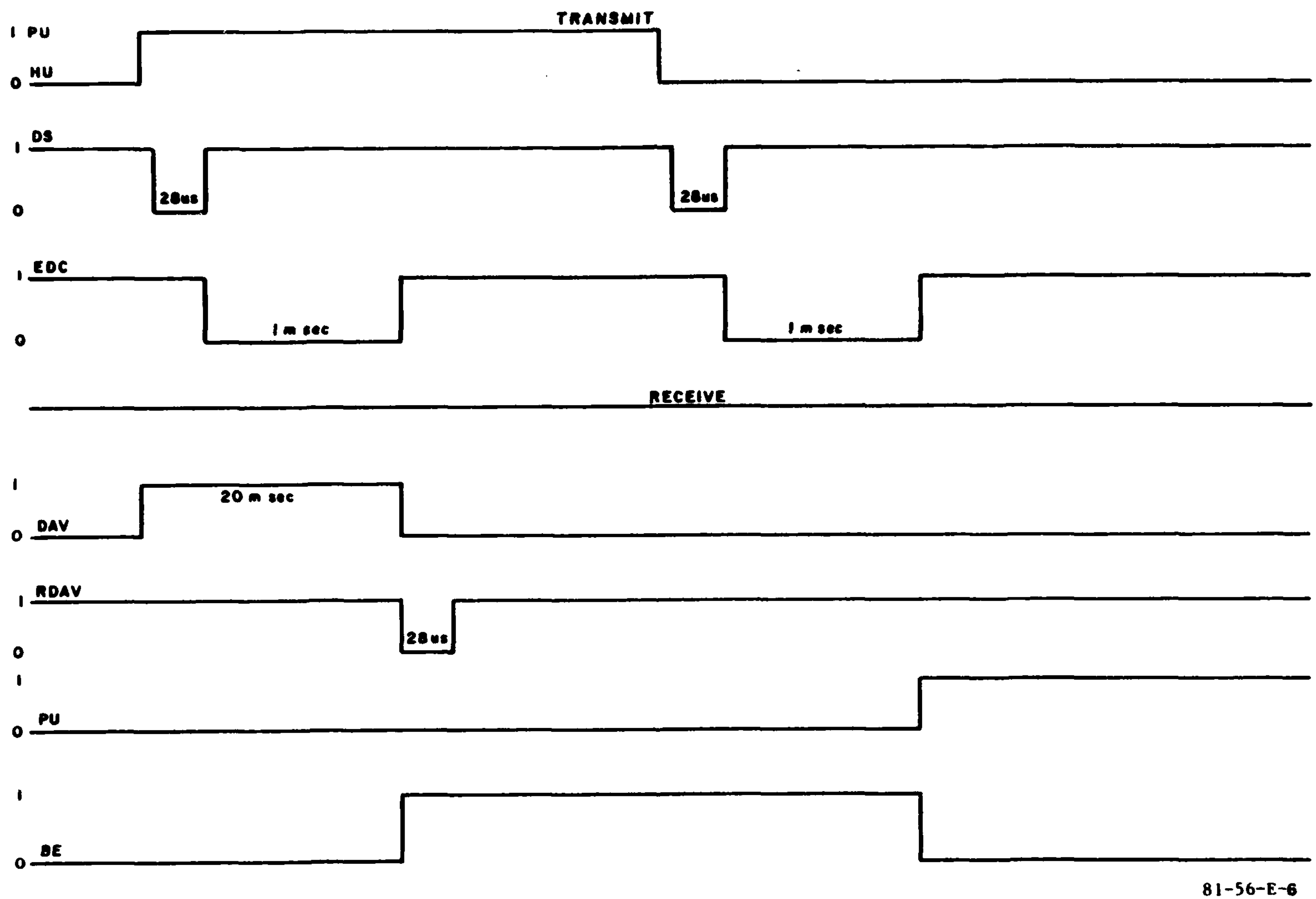
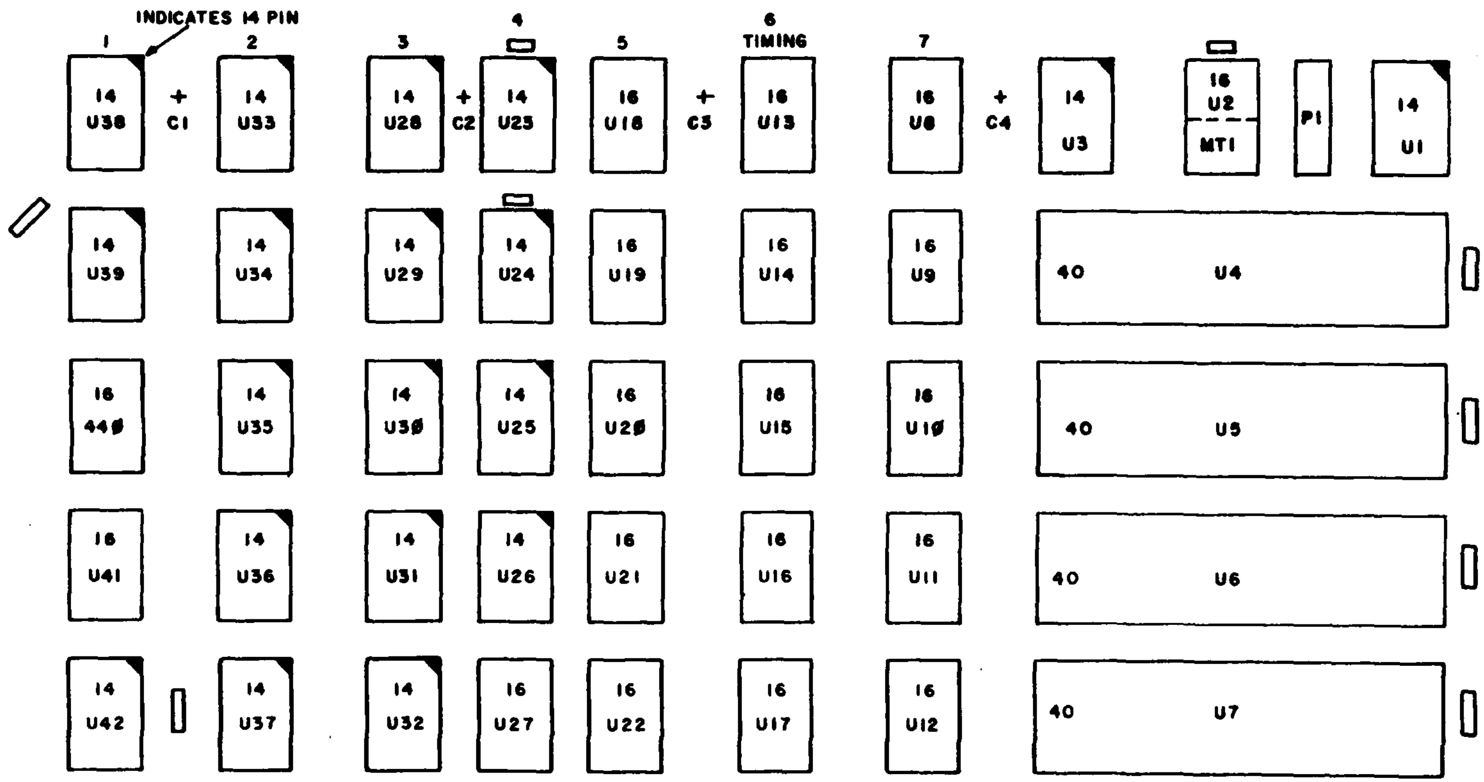


FIGURE E-6.   FOUR-SPECIALIST INTERFACE TIMING



81-56-E-7

FIGURE E-7. STUFFING CHART

APPENDIX F

FAST-FILE RECORDERS

PREFACE.

The two fast-file recorders associated with the Mass Weather Dissemination System Exploratory Engineering Model are used for the filing, closing, and amending of flight plans. Either of the fast-files may be connected to any of the incoming telephone lines in accordance with the concept of one-call service. These units differ significantly from present flight plan recorders in that they allow the pilot the option of reviewing his flight plan and amending it, if required. Each fast-file recorder is capable of recording 108 flight plans without operator intervention.

THE RECORDER.

The fast-file recorders are modified Lanier Tele-Edisette 1400C dictation un equipped with voice operated relay (VOR) circuits. These off-the-shelf items capable of loading 12 standard C-60 leaderless cassette tapes. Each tape store up to nine separate message units before it is automatically replaced w the next cassette in the rack. The number of entries on a tape is opera selectable by a front panel dial. A counter display on the front panel informs the operator of the number of entries on the tape currently in use.

The standard tele-edisette is controlled by a handset, similar to that of an ordinary desk telephone, with capacitive switches embedded near the earpiece. The state of these switches, in the normal operating mode, is decoded into a 4-bit command nibble which governs the operation of the machine. This circuitry was disabled by replacing the comparator integrated circuit (Z102), which interpreted handset commands, with a connector (HC) that supplies a command nibble generated by the mass weather dissemination computer in accordance with the utterance recognition device's (URD) replies. The activation of the recorder, which normally occurred when the handset was removed from its cradle, is accomplished by applying a TTL logic-one level at the input of Z105E on the Lanier control board. Lanier transistor Q104 has been removed.

A solid-state VOR option was also purchased with the recorders. This device enables the circuitry to determine when a speaker has finished his entry. It should be noted that this device is a level rather than syllabic detector and is, therefore, susceptible to extremes in input, which are occasionally encountered when dealing with voice grade telephone lines.

THE INTERFACE.

Communications between the mass dissemination computer, an Interdata 7/32, and the fast-file units are asynchronous. Data are transmitted from the 7/32 processor by means of a programmable asynchronous line module (PALM) I/O port. A custom interface located within the fast-file unit, designed around a General Instruments AY-5-1013 Universal Asynchronous Receiver Transmitter (UART), interprets the 7/32 processor commands and presents them to the command nibble of the fast-file unit. The interface also provides the 7/32 processor with pertinent status

information regarding the fast-file unit. The physical communications link between computer and fast-file is in accordance with Electronics Industries Association (EIA) standard RS-232C. Transfer occurs at 1200 baud with 8 data bits, 2 stop bits, and no parity.

The use of an asynchronous control link, rather than the more common parallel method, presents a number of advantages. The fast-file units may be located remote from the 7/32 computer in the operational area of the Flight Service Station without requiring extensive buffering. The PALM channels of the Interdata 7/32 are designed for the control of remote data entry terminals. Since the fast-file interface is designed in such a manner as to be analogous to such a device, no additional software device handlers are required to control the fast-file. No additional hardware is required on the 7/32 computer side of the link to complete the interface. The fast-files may be controlled by virtually any computer that is capable of supporting remote terminals.

FAST-FILE OPERATION.

The fast-file recorders are capable of accepting six basic single byte commands from the 7/32 computer. These are: sense status (X'40'), seize/dictate (X'30'), rewind/listen (X'25'), seize (X'20'), end (X'2E'), and unseize (X'00').

The normal operational sequence of commands is: sense status, seize/dictate, seize, end, rewind/listen, and unseize. The rewind listen sequence is an option given to the pilot and is not utilized in every flight plan transaction. The caller also has the option to amend his flight plan, in which case the command sequence following the sense status command and prior to the unseize command may be repeated until the caller is satisfied with his entry.

All commands to the fast-files are single-byte transfers. Detection of a command by the UART causes DAV1 to go high. This transition activates a one-shot (U8A) of duration 50 milliseconds which provides a data settling period and command sequence initialization. During the period of U8A, the reception of further data by the interface is inhibited by logically "oring" any incoming signal with a logic-one causing the input line of the UART to remain at mark. Completion of U8A's period fires one-shot U8B which resets the UART DAV1 to zero. This pulse also serves to provide the external command strobe (TRIGO) to the Lanier control logic.

When the mass weather dissemination computer's monitor routine receives an URD response representing a request for a fast-file, it selects from its table of available devices the first fast-file unit not being utilized by the system. The monitor routine then issues the sense status command to determine if the fast-file unit is unavailable due to external operator intervention. The low-order nibble (X'0') of the sense status command is used to clear the command latch (U11). Bit six of the command causes the Q output of U8A to act as a transmit stobe to the UART which sends a status byte to the 7/32 computer. The return of any code, except X'08', is interpreted as bad status and causes that unit to be removed from the available device list until a code of X'08' is received. The next available fast-file, if any, is then polled in the same fashion. If no fast-files are available, the requester is put onto a wait list until one becomes available. If a fast-file is not on, or the on-line/off-line switch (53) is in the off position at sense status time, the RS-232C parameter (data set ready) is at 0 volts. This results in a time-out code of X'82' to be generated in the PALM driver and

returned to the monitor routine. During the transmission of any data to the 7/32 processor by the fast-file, the input line to the UART is disabled by logically "oring" the complement of UART signal EOC1 with the incoming serial data. This is done in order to inhibit the data wraparound feature (echoplex) of the PALM.

Once it has been determined that a fast-file is available, that unit must be connected to the requesting caller and the recording sequence begun. This requires the issuance of two commands by the monitor routine. The first command must establish the audio link between the caller and the fast-file by means of the crosspoint switch array. Secondly, the fast-file recorder must now be instructed to begin the record sequence. This is accomplished by sending the fast-file a seize/record command, X'30'. The fifth bit of this command causes the fast-file to be seized by the 7/32 computer, by strobing a logic-one into the Q output of flip-flop U10A. This output is then applied in lieu of the collector voltage of Lanier transistor Q104. At this time, the command nibble is X'0'. Bit four of the seize/record command triggers one-shot U15A for a period of approximately 0.4 seconds. During this period, a ready tone is generated by the Lanier circuitry. At the end of 0.4 seconds, the ready tone is turned off and one-shot U15B is triggered. The resulting pulse toggles the most significant bit of command latch U11 to the logic-one state. This results in a new command nibble of X'8', which is the dictate command. Transportation of tape through the fast-file may begin at this time or at the first detection of voice by the VOR circuit of the Lanier. This option is controlled by the setting of interface switch S1G. In the open position, tape transport begins immediately.

Activation of the VOR circuit by the dictate command induces a negative-going transient which sets the output of the independent J-K flip-flop of U5 to the logic-one state. At this time, a time-out circuit is activated. This circuit consists of an astable oscillator (U4B) with a period of 5 seconds and an 8 to 1 duty cycle driving a 3-bit counter. If no speech is detected by the VOR circuit in approximately 20 seconds, the counter reaches X'4', which triggers one-shot U18B, generating a 28 microseconds pulse which causes the time-out code (X'C8') to be transmitted to the 7/32 computer. If the VOR circuit is not implemented, the output of U6A (BEEPO) can be used to reissue the cue tone every 5 seconds until time-out occurs. This would necessitate a fixed time interval for flight plans which would be less flexible than the method currently employed. The reception of the time-out causes corrective measures to be taken by the monitor routine. Detection of speech by the VOR inhibits time-out oscillator U4B before the time-out condition is reached.

The pilot now speaks his flight plan into his telephone instrument. When the pilot completes his entry to fast-file, he remains silent for approximately 5 seconds. This causes a negative-going pulse to be presented to the input of the independent flip-flop of U5. The negative transition of the output of U5 triggers one-shot U12B which causes the transmission of the operation complete code (X'48') to the 7/32 computer. The monitor software now breaks the audio link to the fast-file and transmits the seize command to the fast-file. This command sets the command nibble to zero while maintaining the fast-file in the ready state. The seize command also puts an inaudible mark on the tape to prevent playback beyond the recorded portion of the tape.

The monitor software must now determine if the pilot wishes to review his flight plan. If the pilot does not wish a replay of his entry, the fast-file is issued the unseize command. This causes an inaudible end of record mark to be placed on the tape to isolate the current entry from future entries, the fast-file front panel counter display is incremented, and the unit is returned to available device status.

If the monitor software has determined that the user wishes to review his last entry, the end command is transmitted to the fast-file. This command causes an end-of-record mark to be placed on the tape but does not cause the counter to be incremented or the fast-file to be unseized. The audio link between caller and fast-file is then reestablished. Finally, the 7/32 computer issues the rewind/listen command to the fast-file. This causes X'5' to appear at the output of command latch U11. The fast-file now rewinds to the beginning of its last entry. At the end of the rewind operation, the signal REYC1 goes high. This signal fires one-shot U13B, causing a 28US low pulse at the complimentary Q output. This pulse clears D flip-flop U10B, thereby inhibiting the transfer of the LSB of the command nibble to header control (HC). This pulse also causes TRIGO to go low to provide a command strobe so that the Lanier control logic will accept the new command nibble, X'4'. This is the listen command.

Completion of the listen sequence occurs when the Lanier circuitry detects the ready mark created by the seize command. Completion of this operation causes signal LISTEN1 to go low causing one-shot U13A to fire, thereby transmitting the operation complete code, X'48', to the 7/32 computer. It should be noted that signal LISTEN1 is prone to transitions when the fast-file is not in the rewind/listen state. Therefore, the operation of one-shot U13A is inhibited when the LSB of the command latch is at the logic-zero level.

The monitor routine now determines if the pilot wishes to amend his last entry. If the pilot's response is negative, the fast-file is unseized as previously described. If it is determined that the pilot wishes to amend his entry, the sequence previously described is repeated, beginning with the establishment of the audio link between caller and fast-file. Previous entries to fast-file made on the same call are not replayed during subsequent review sequences, due to the presence of the end-of-record mark created by the end command.

As a precautionary measure, if a tape transport error occurs during a fast-file sequence, the signal JAM0 goes low, triggering one-shot U12A. This results in error code X'40' being transmitted to the 7/32 computer so that corrective action may then be taken by the monitor software. No tape transfer errors have occurred in normal operation since the incorporation of the fast-file into the Mass Weather Dissemination System.

HARDWARE NOTES.

The fast-file interface is located on a single card mounted behind the fast-file control board. The 5-volt power supply required for the interface circuitry is obtained from the 5-volt supply of the Tele-Edisete. The plus and minus 12-volt supplies required by the UART and RS-232C line drivers are supplied by an independent Acopian plus and minus 15-volt supply which is reduced to the proper level by the Zener diode arrangement located on the power header (HP). Power input to all components of the interface are decoupled by 0.01UF capacitors.

Signals described in the fast-file schematics use the same logic convention as employed in the Interdata documentation. A signal followed by a 1 is active in the logic-one state; e.g., the signal LISTEN1 indicates that the fast-file is in the listen mode when that signal is high. Conversely, a signal followed by a 0 is low active; e.g., JAM0.

The telephone audio inputs provided to the fast-file units are customer tip (CT) and customer ring (CR). This pair is isolated from the CDH interface by a 600-ohm bridging transformer located external to the fast-file units. The audio connection is made at pins five and six of transformer T201 on the Lanier audio board. Lanier resistor R209 is disconnected from the collector of transistor Q202. Input levels to the fast-file are maintained by an automatic level control circuit on the audio board. Audio playback level is controlled by potentiometer R221. Input trigger level and delay time for speech detection are controlled by two potentometers located on the VOR board.

MNEMONICS

BEEP0	Auxiliary cue tone control
BLV1	Busy level voltage — Lanier signal
CC0	Condition change — transmits status to the 7/32 computer
CCI0	Condition change inhibit — inhibits CC0
CS0	Command strobe — loads command latch U11
DAV1	Data available — command byte received
HC	Header control — replaces Lanier IC Z102
HF	Header feedback — sense connections to Lanier circuits
HIO	Header input output — used for RS-232C connections
HP	Header power — mounts power regulation circuits
HT	Header timing — mounts R-C timing elements
LOG'1'	Logic 1 signal — +5 volts through 1K ohm resistor
P1	Potentiometer — used for UART clock
PS	Acopian power supply — +/-15 volts, 100MA
REVC1	Reverse complete — end of rewind operation
RI0	Reverse inhibit — modifies rewind command to listen
RTE1	Ready tone enable
SIN	Serial input — RS-232C
SOUT	Serial output — RS-232C
TRIG0	Trigger — strobe to transfer command nibble
TO1	Timeout — no speech detected by VOR circuit
VOR	Voice operated relay — solid state

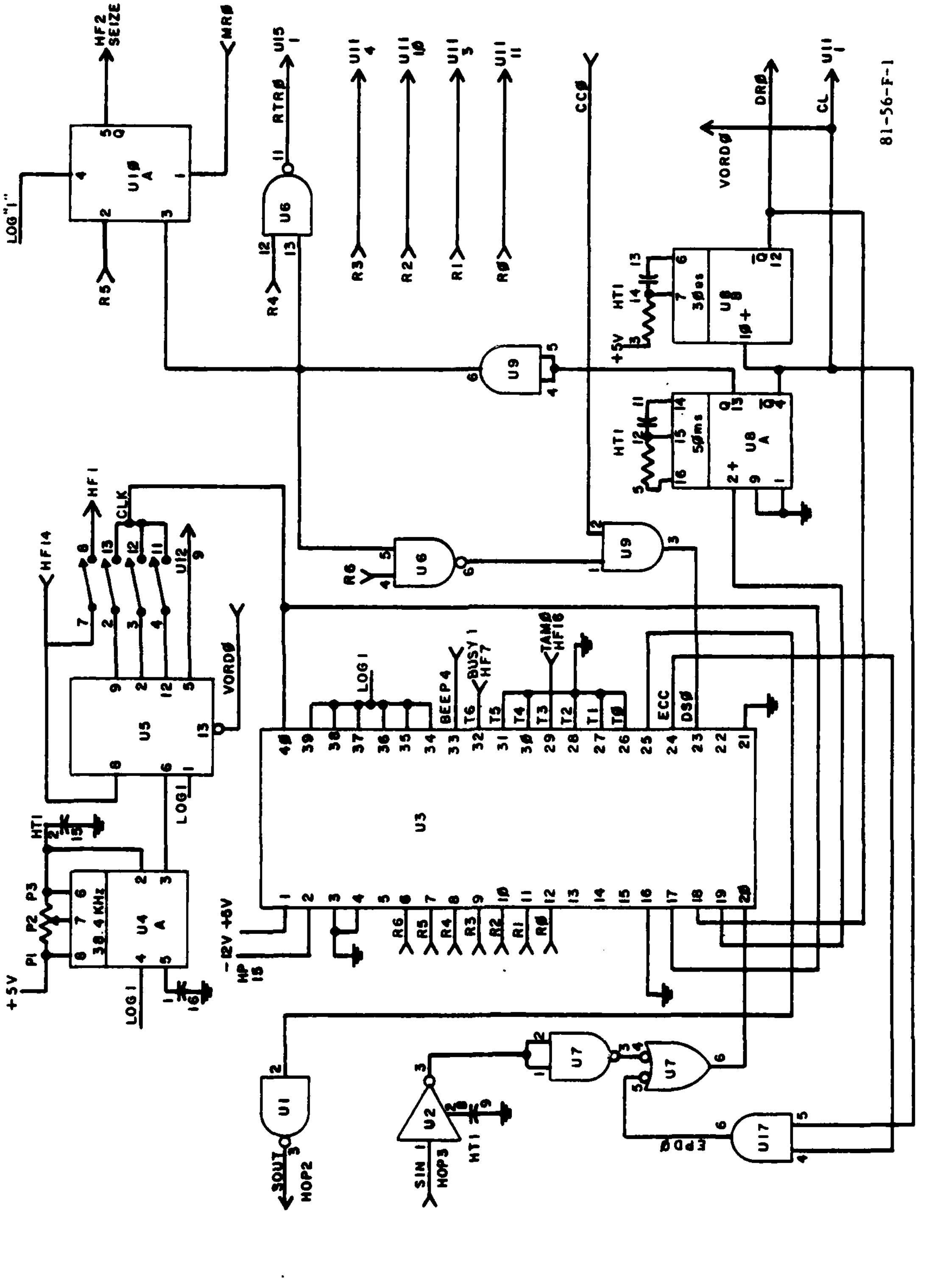


FIGURE F-1. INTERDATA — LANIER INTERFACE (SHEET 1 OF 4)

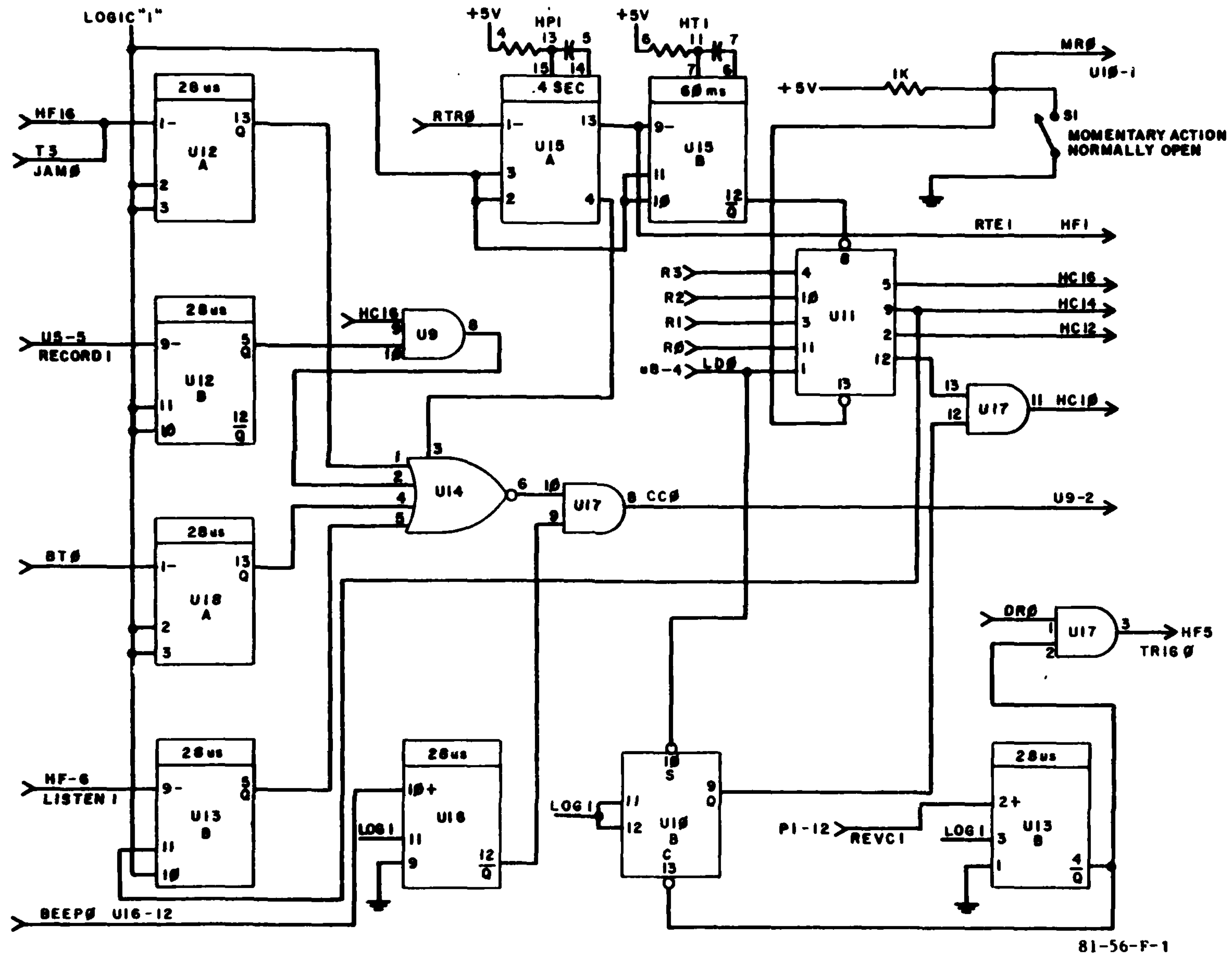
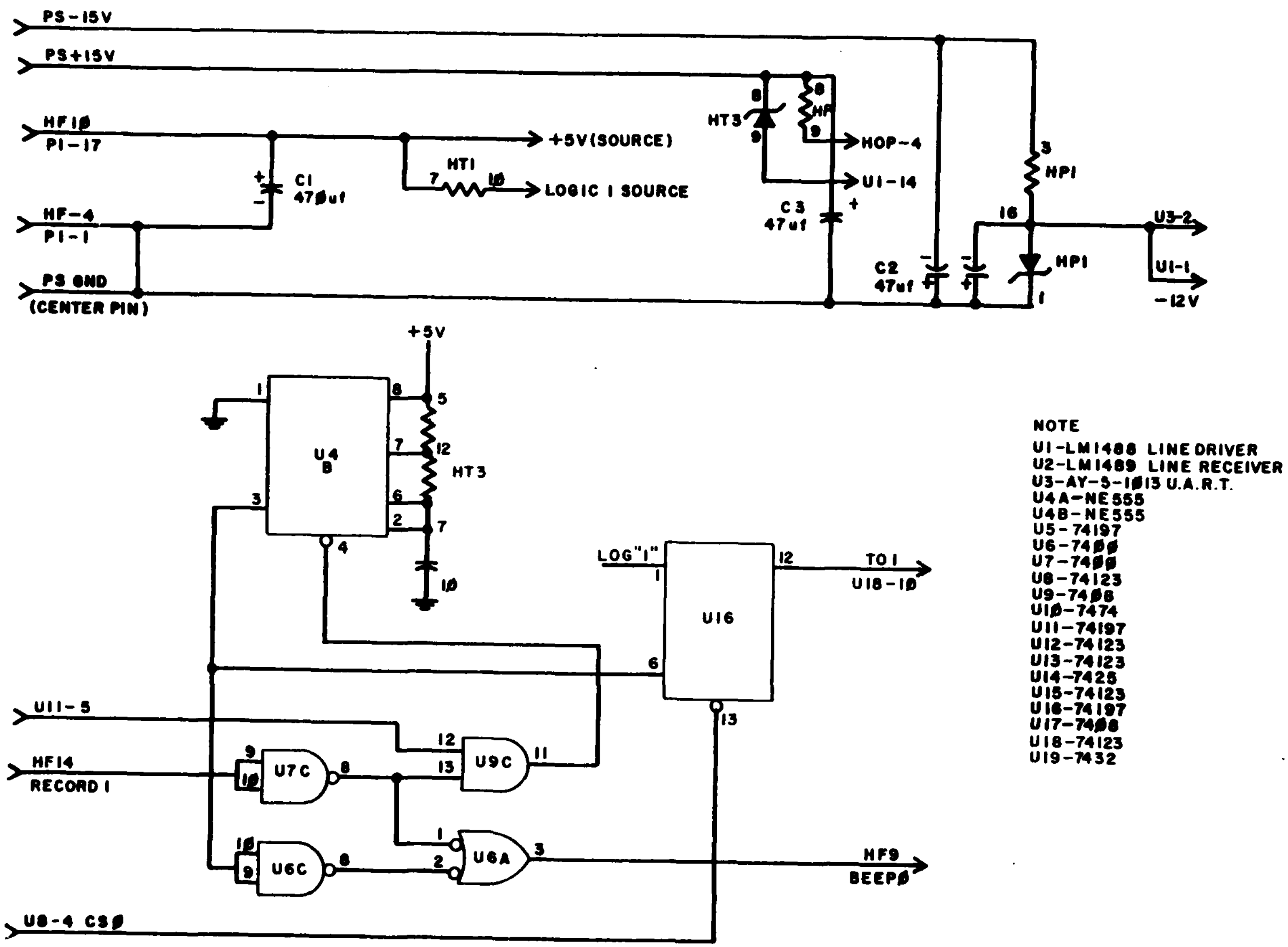


FIGURE F-1. INTERDATA — LANIER INTERFACE (SHEET 2 OF 4)



81-56-F-1

FIGURE F-1. INTERDATA — LANIER INTERFACE (SHEET 3 OF 4)

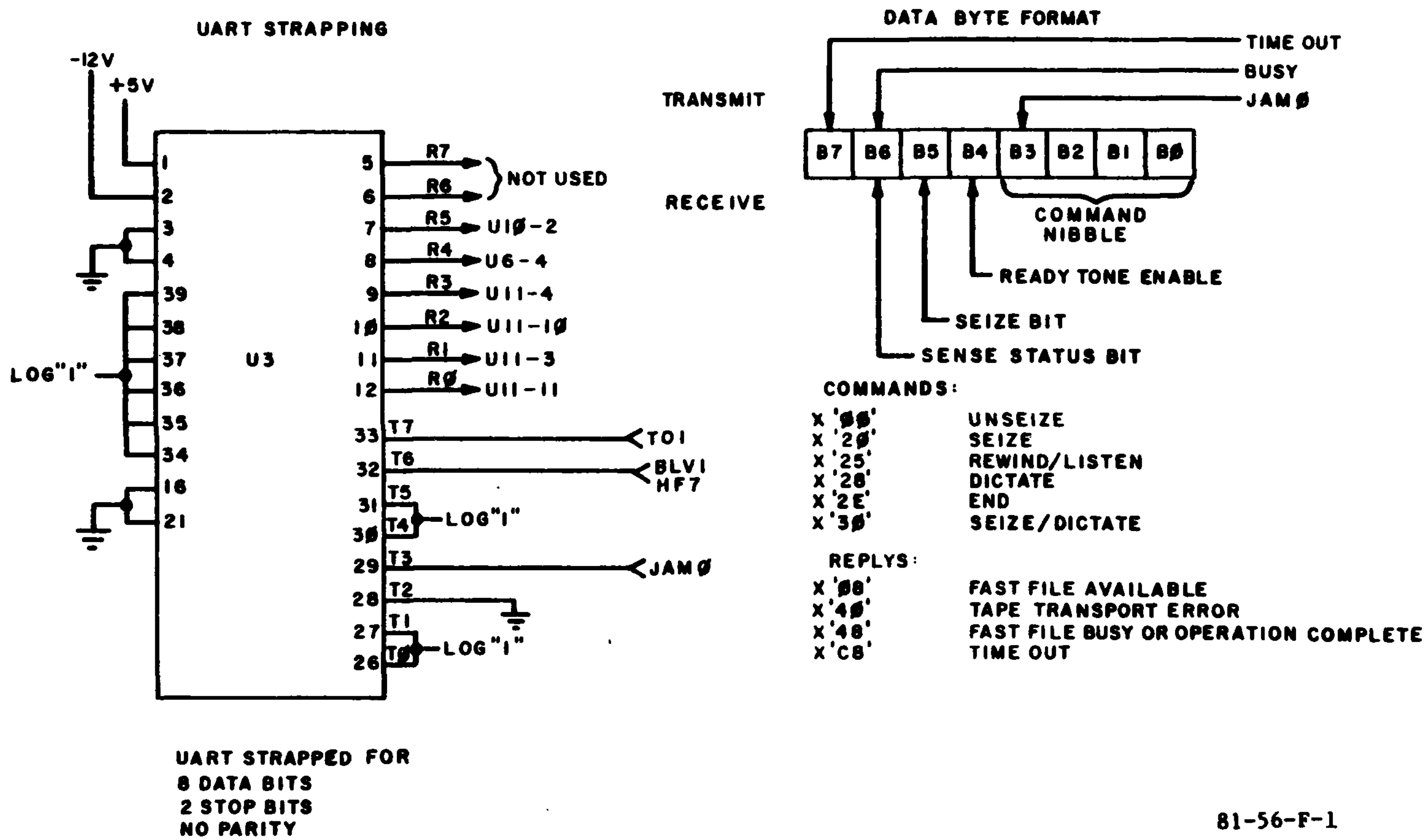
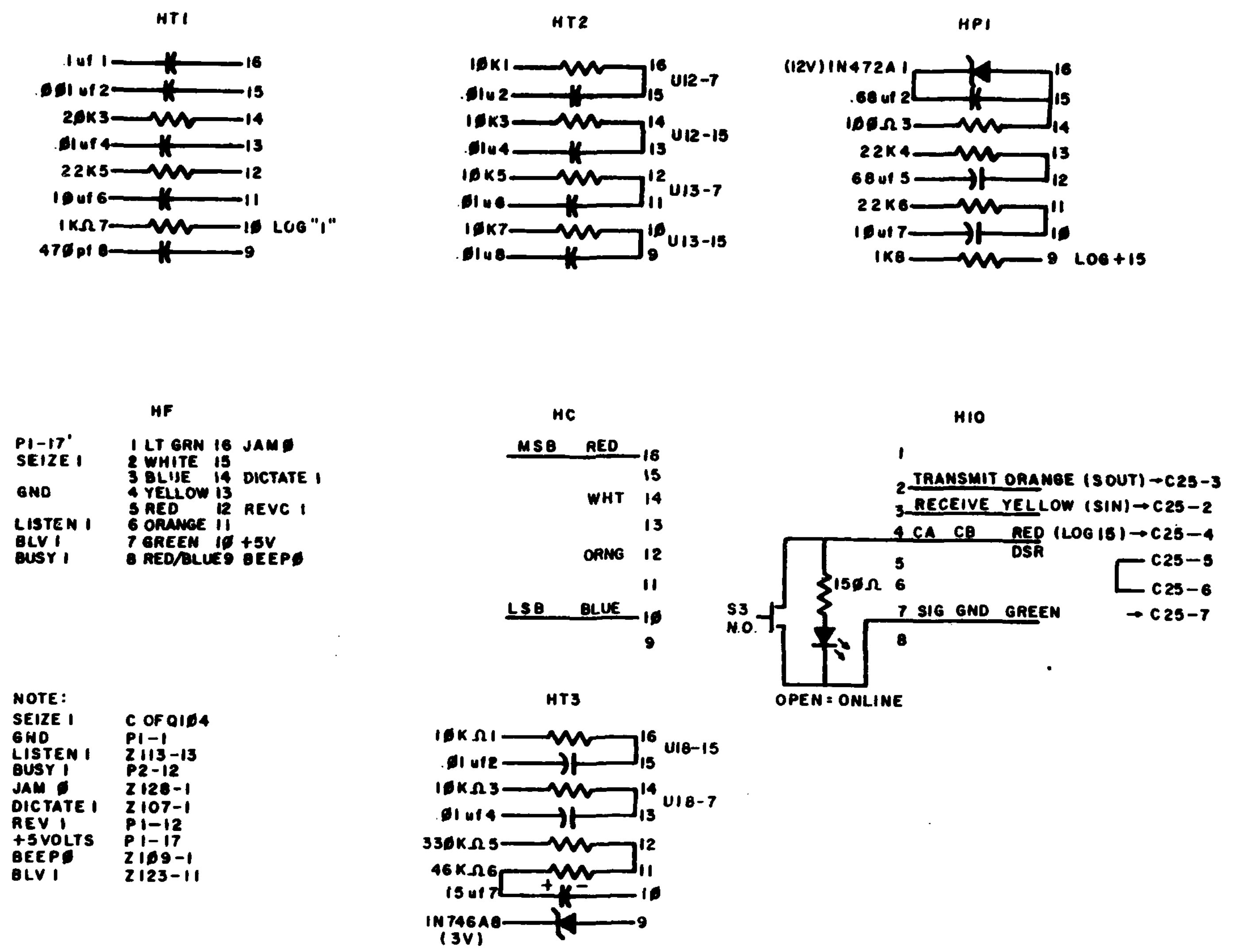
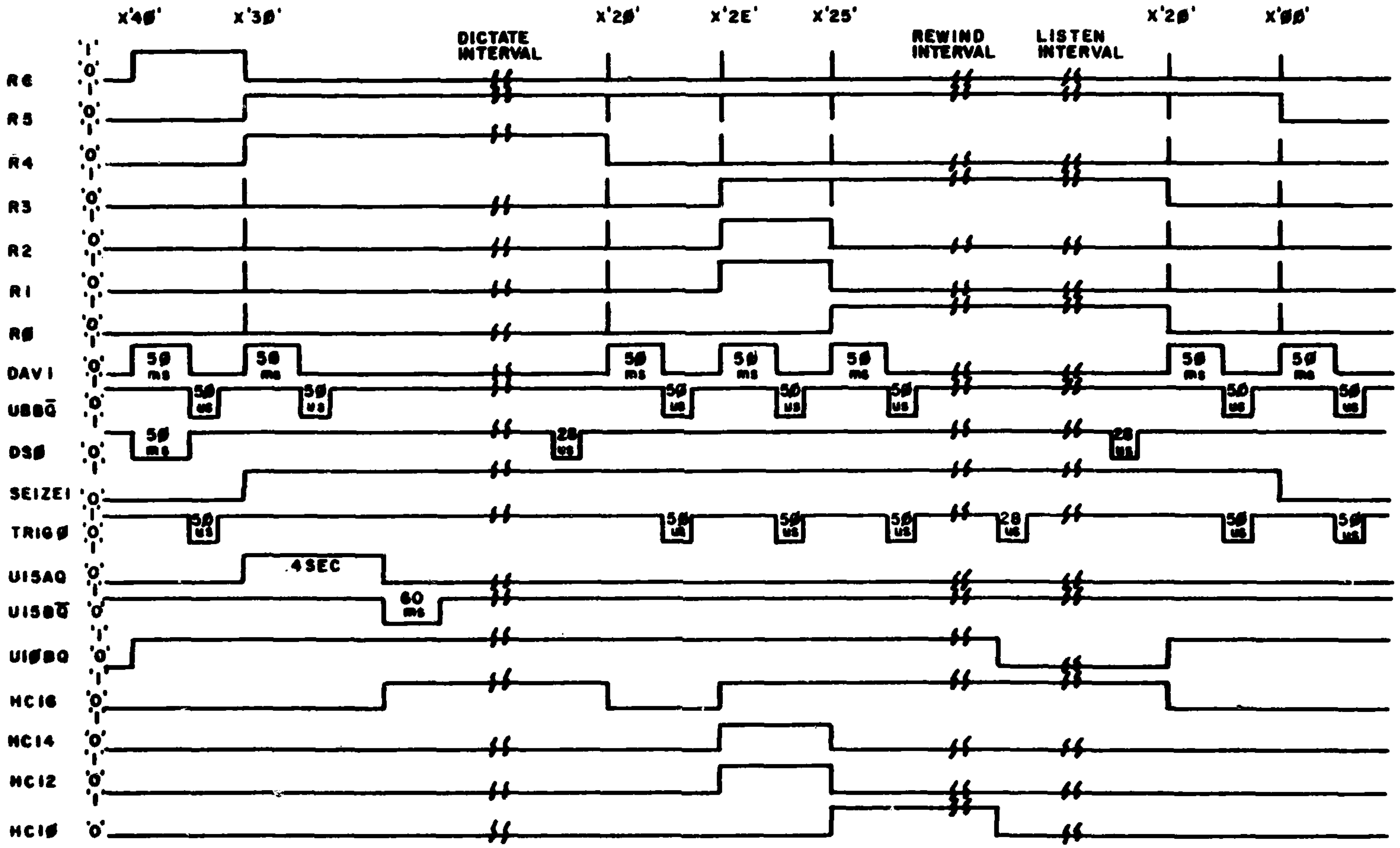


FIGURE F-1. INTERDATA — LANIER INTERFACE (SHEET 4 OF 4)



81-56-F-2

FIGURE F-2. INTERDATA — LANIER INTERFACE (HEADERS AND CONNECTORS)



NOTE 1: PULSE DURATIONS INDICATED ABOVE ARE NOT DRAWN TO SCALE. USE FOR TIMING RELATIONSHIPS ONLY.

NOTE 2: THE SEQUENCE SHOWN ABOVE IS AN EXAMPLE ONLY. VARIANCES MAY OCCUR DEPENDING ON USER REPLIES.

81-56-F-3

FIGURE F-3. INTERDATA — LANIER INTERFACE TIMING

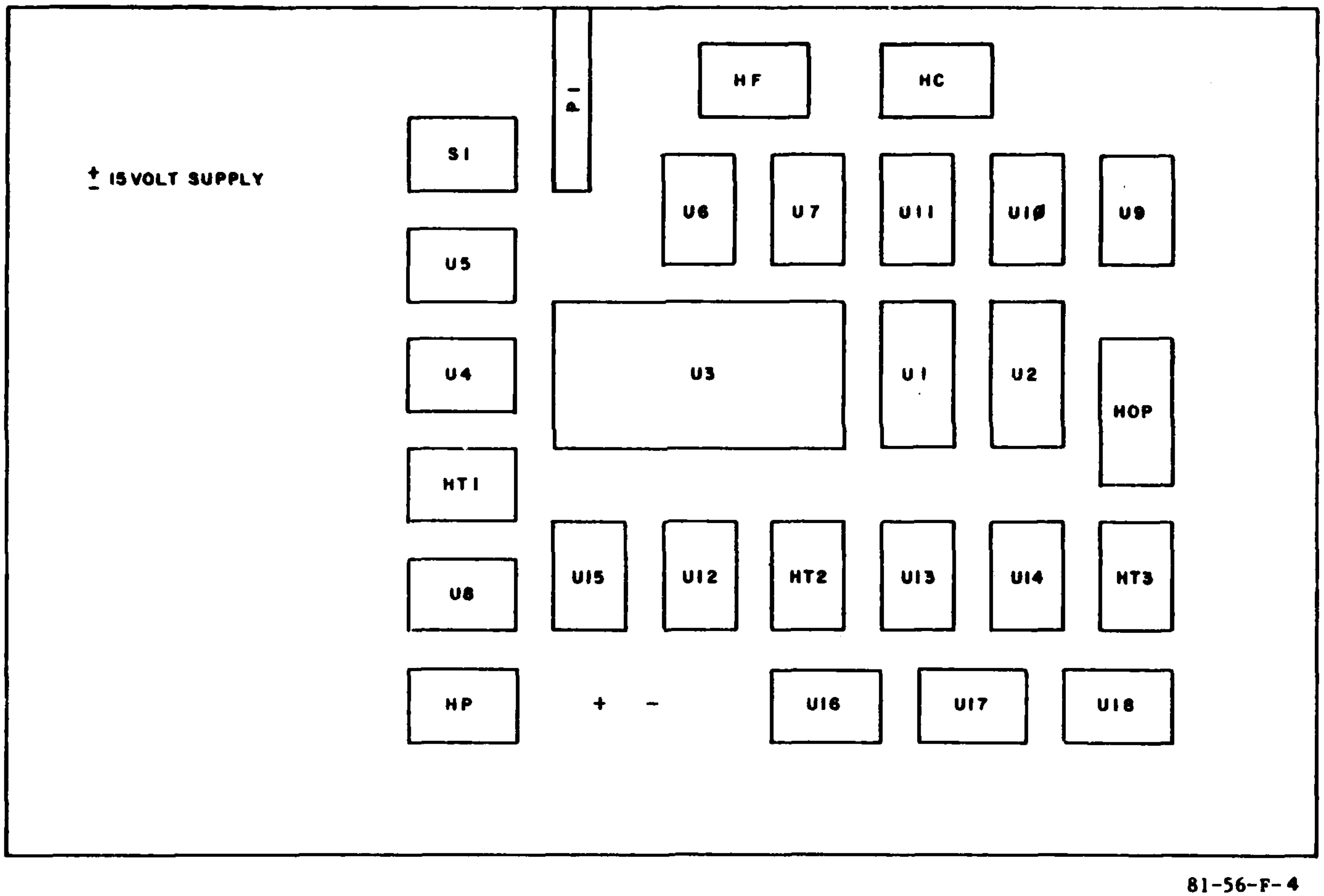


FIGURE F-4. INTERDATA — LANIER INTERFACE (STUFFING CHART)

		Output Pin#s of Z 102 (3302)			
Command Nibble		14	1	13	2
Ready		0	0	0	0
Dictate		1	0	0	0
Reverse		0	1	0	1
Listen		0	1	0	0
Instruction		1	0	1	0
End		1	1	1	0
Fast Forward		1	1	0	0
Intercom		1	1	1	1
HC quivalents	Pin#	16	14	12	10

FIGURE F-5. HEADER CONTROL EQUIVALENTS

HF	Control	Function
1-White	P1-17' (cut)	VOR to control logic
2-Black	P1-8	Seize (Q104 removed)
3-NC		
4-Green	P1-1	Ground
5-Black	Z104-9	Trig0
6-Blue	Z113-13	Listen1
7-Red	Z123-11	Busy level voltage
8-Black	P-12 trace	Busy1
9-Yellow	Z10-1	Beep0
10-Black	P1-15 trace	+5v (supply to interface)
11-NC		
12-Red	P1-12	Rev1
13-NC		
		Dictate 1
14-Black	P1-17	VOR to interface logic
15-Black	P1-8	Ready tone disable
16-Brown	Z128-1	Jam0

FIGURE F-6. HEADER FEEDBACK CONTROL INTERCONNECTIONS

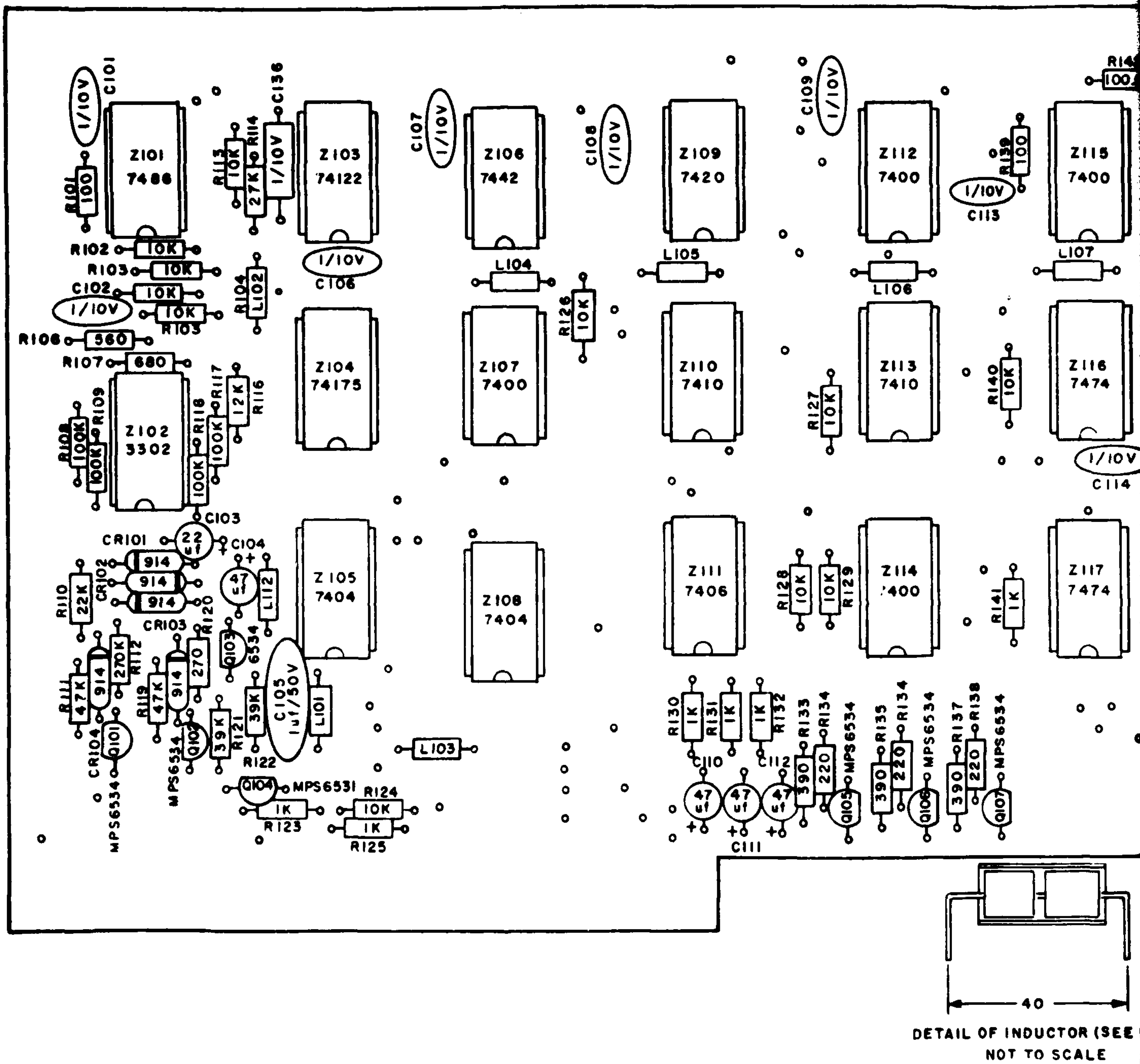
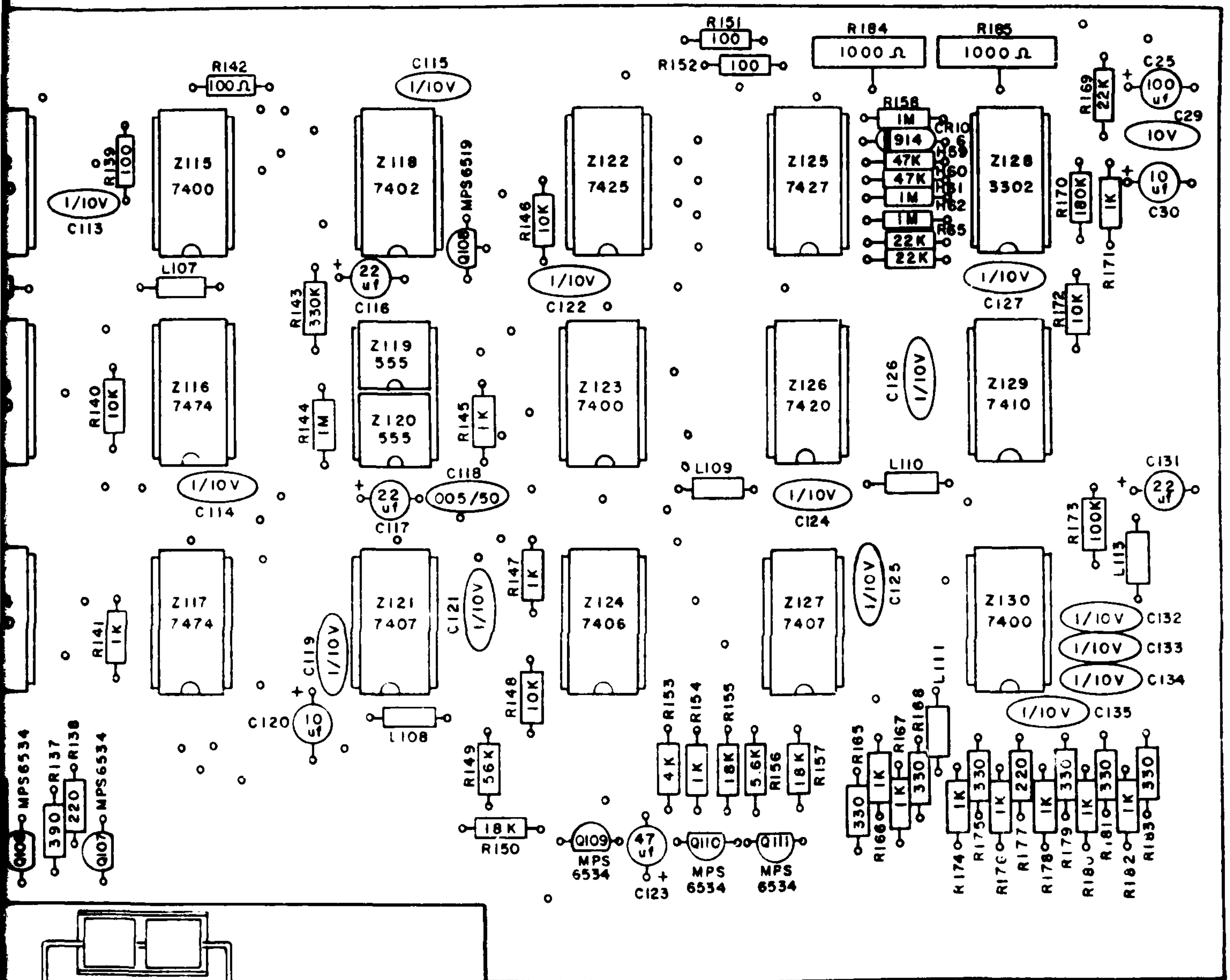
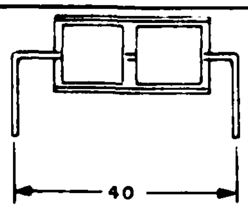


FIGURE F-7. 1400C CONTROL BOA





NOTE:
1. ASSEMBLE INDUCTOR AS SHOWN
INSERT INDUCTOR INTO ALL POSITIONS
DESIGNATED "L"

DETAIL OF INDUCTOR (SEE NOTE 1)
NOT TO SCALE

R184–END OF CASSETTE SENS

81-56-F-7

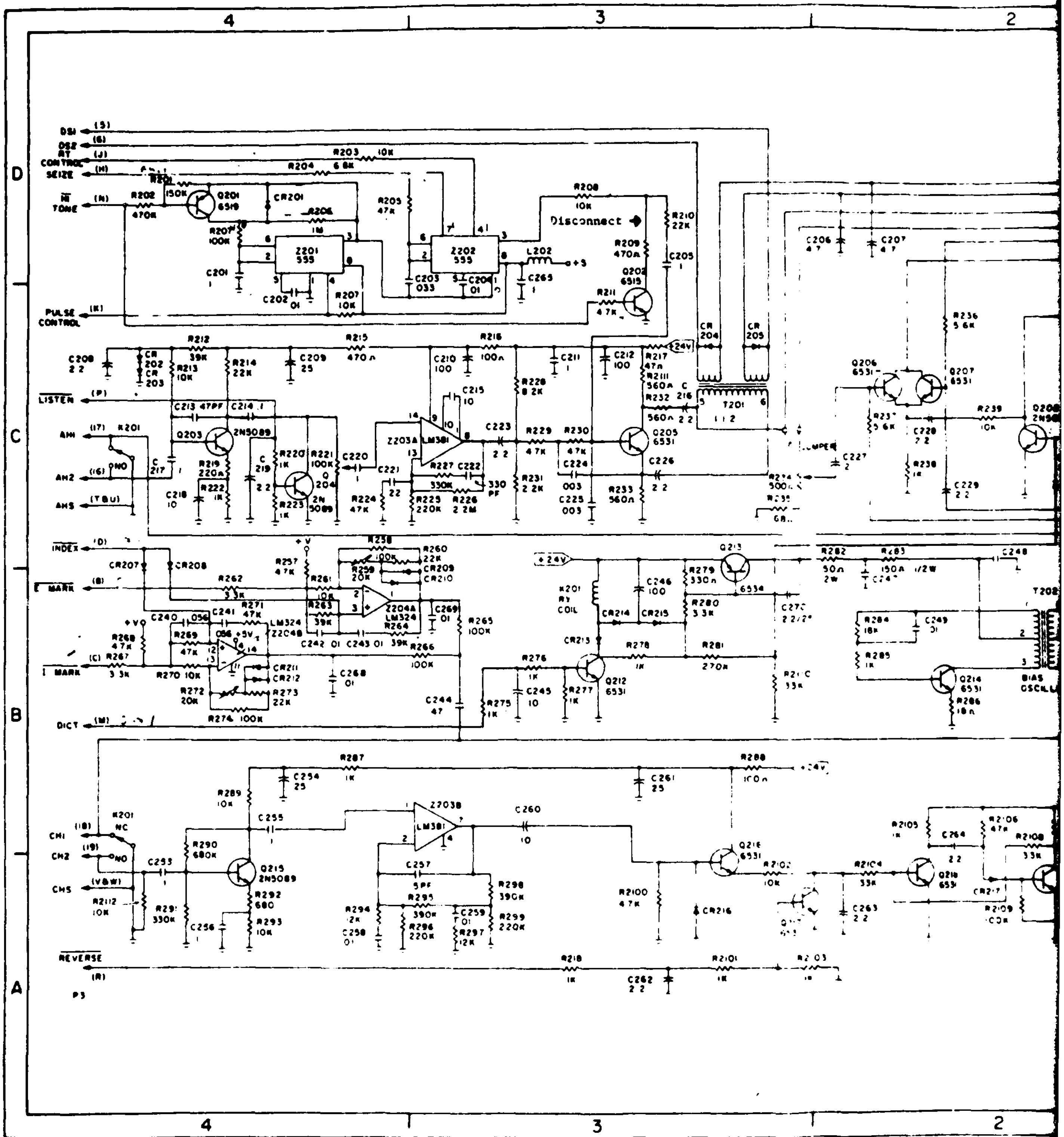
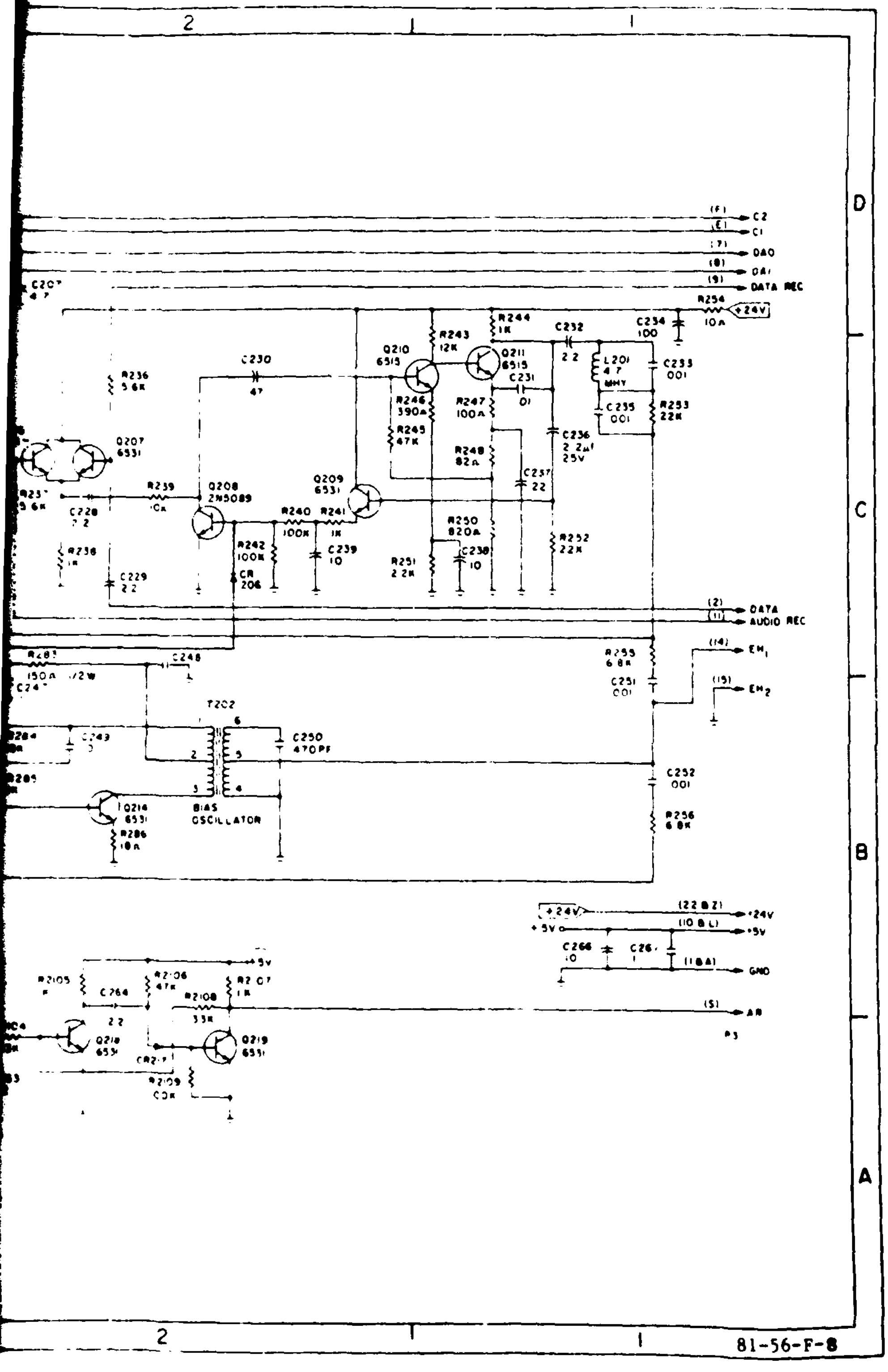


FIGURE F-8. AUDIO BOARD SCHEMATIC 1400C



SCHEMATIC 1400C

2

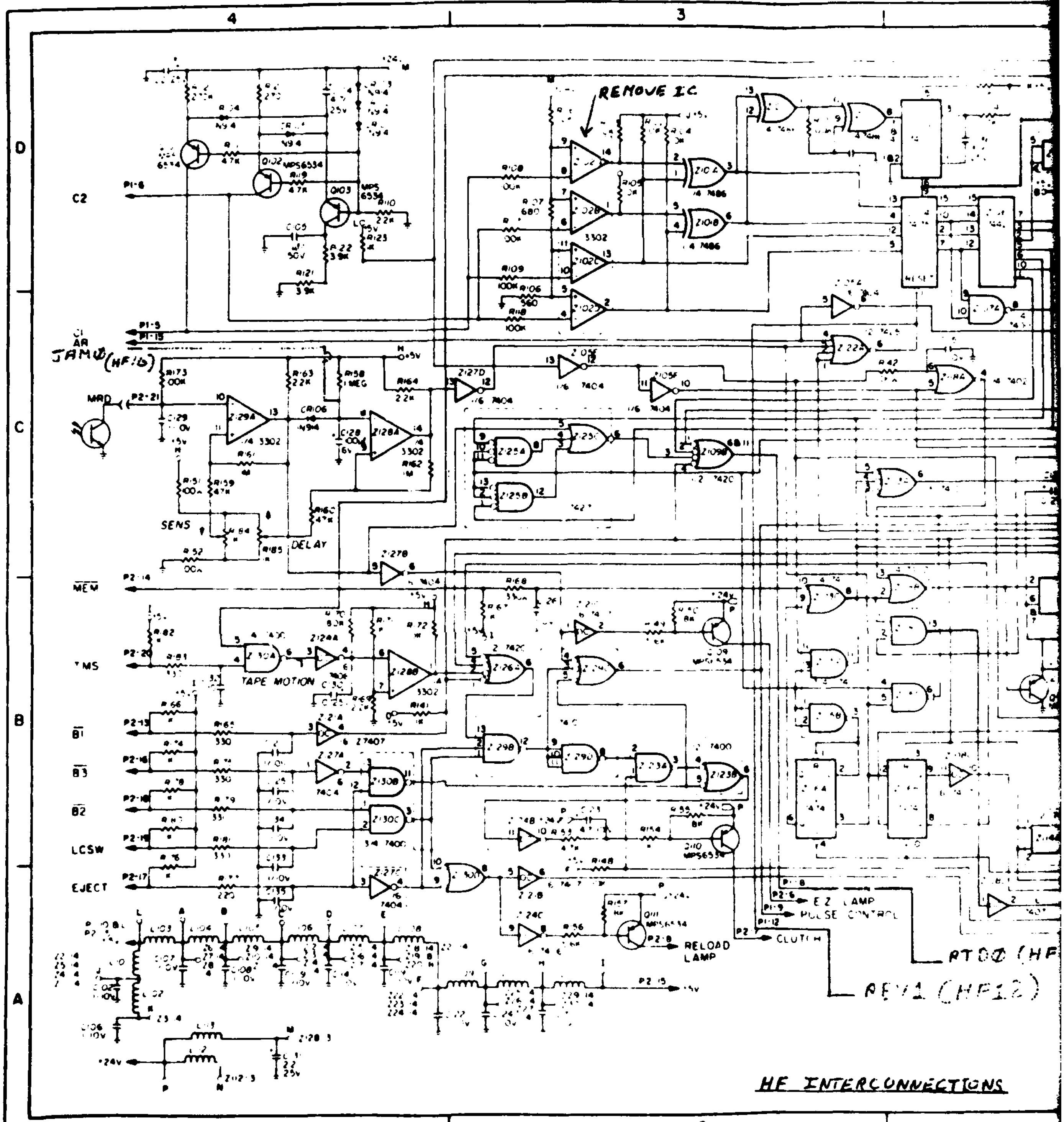


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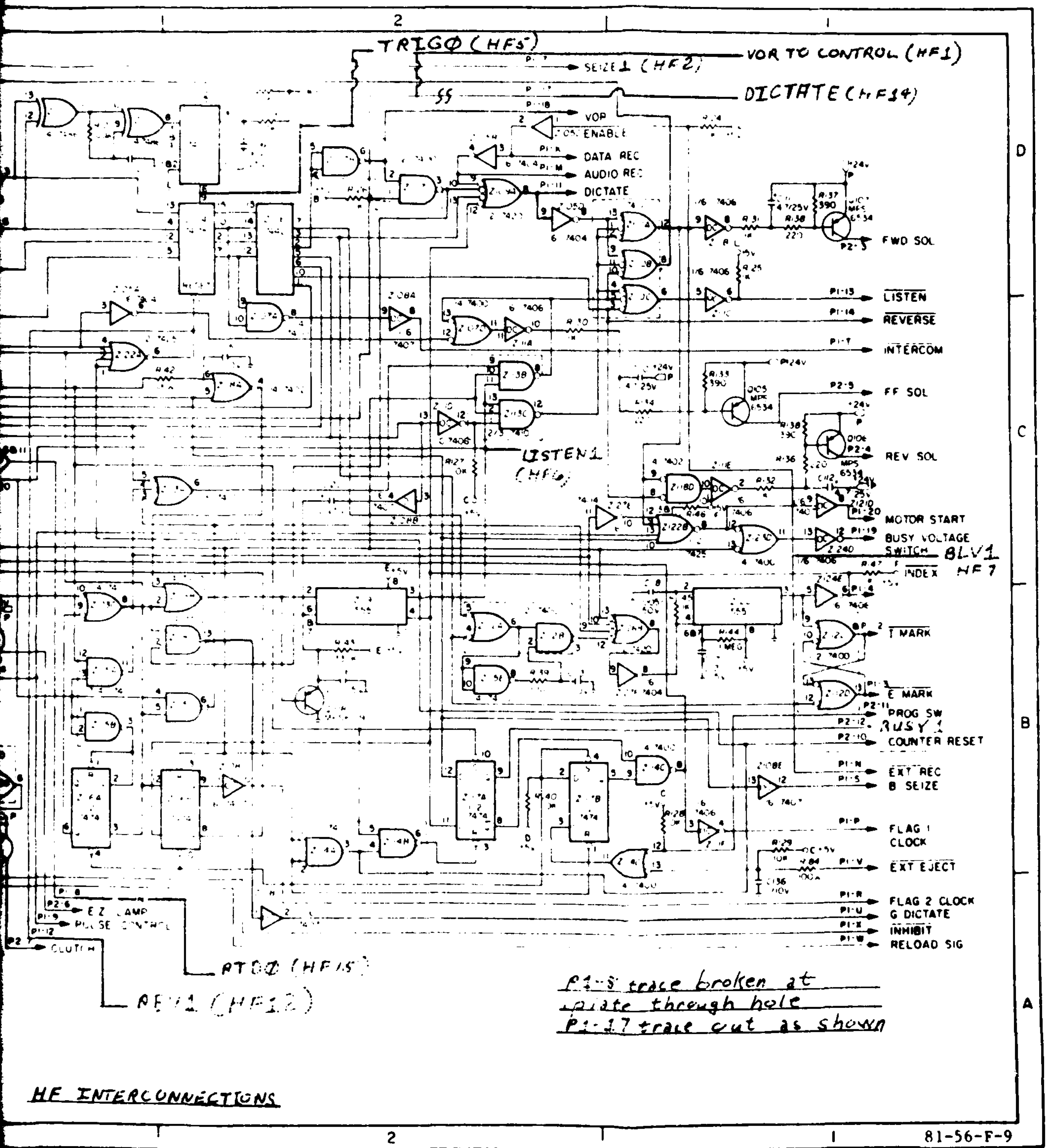
FIGURE F-9. 1400C CONTROL BOARD



S146

F-9. 1400C CONTROL BOARD SCHEMATIC

2